EEL 4915
Senior Design II
Unplugged: Solar Powered Audio Amplifier with DSP Effects

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Executive Summary

Musicians and music enthusiasts are always looking for new ways of sharing their craft with other people. Sound recording technologies such as magnetic tape and compact discs allowed them to deliver their recorded performances to wider audiences. Still, live music performances are an irreplaceable part of a musician's life. Musicians that use electrical instruments such as keyboards and electric guitars are at a disadvantage with respect to their acoustic counterparts since they depend in having electronic amplifiers and being attached on the power grid to play. A solar powered amplifier would give them more freedom to play anywhere they wanted too without being tied to the regular power grid. Possible uses of this device could be as personal amplifiers for street performers, music system for outdoor parties or a public address system for small campsites or any other kind of public event where connecting to the power grid is difficult or impossible. This is the main goal of the "Unplugged" sound system.

The motivation to take on this enterprise comes from the personal desire of team members to be able to play musical instruments or just enjoy music from a laptop computer or iPod without the need of running long and cumbersome extension cords. The design for the Unplugged sound system includes one input for either a low impedance microphone, or a musical instrument such as electro-acoustic guitars and an 1/8 inch connector capable of interfacing a laptop computer or MP3 player. The system's versatility is further enhanced with the inclusion of the BTSE-16FX DSP effects module. This module will let the user implement one of sixteen different audio effects in the audio source of their choice. At the core of this design is the TDA7396 amplifier chip which is capable of delivering a power output of 45 Watts RMS into a 4 Ω speaker. The whole system is powered by a 12 Volt 5 ah SLA battery that is trickle charged by a 20 Watt photovoltaic cell.

1 Technical objectives and requirements

There are three well-defined subsystems within the *Unplugged* sound system. These are: audio amplification, digital signal processing and Power collection/management subsystems. These posses different but complementary technical objectives such as:

1.1 Audio Amplification Subsystem

-The device will be capable of driving 2 different audio sources for a period of 3 minutes or more.

-Each analog audio input will be equipped with a two-tone control circuit with a dynamic range of +/- 6 dB to compensate for any acoustical inadequacies of the environment where the sound system is used

-Each individual analog input must posses a dedicated level control.

-Should weight less than 30 pounds

-The system will include a ¼ inch TRS input for a microphone or instrument level signals and an 1/8 inch input for an iPod or auxiliary audio source.

1.2 Digital Signal Processing Subsystem

-The user must be able to pick between audio effects of digital reverbs, delays and halls and in which analog audio input they will act.

-The latency between the implementation of the audio effect and the user's input must be minimal since the device is meant to be used in live performance.

-The user must be able to regulate how loud the audio effect is with respect to the clean audio input being applied.

1.3 Power Management System

-The system must be capable of fully charging the SLA battery for optimal operation of al peripheral devices.

-The system must be able to run off a 12V battery.

-The battery must be charged solely by a 20W solar panel

1.4 Monitoring Subsystem

-User must be able to monitor parameters such as battery voltage, solar panel voltage and state of charge of the battery.

-User must be able to monitor which effects is applied to the microphone/ instrument input.

-The measurements for solar panel and battery voltage must be accurate to the nearest tenth of volt.

2 Technology Research

2.1 Previous Works / Similar Projects

In this section we would like to discuss both retail products and student projects that have been found as a result of research for this project. Many projects have been found that relate to this project. In the wide array of similar products, we found three to be more relevant with the rest. These three previous works will be discussed below.

2.1.1 Similar Projects

This project is a very basic student project. The amplifier is charged through solar panels, and it is meant solely to work outside, as it has no battery. The student was able to build the project during a summer break as it is described on the website. The student designed his own amplifier circuit using an amplifier chip, and also designed the circuit in order to charge this amplifier circuit through the solar panels. As this project is very basic, no sound effects of any type were used. This project also used a PCB sold at Radio Shack. The project box is a set of two old speaker boxes that were transformed in order to hold the amplifier circuit and the solar panels.

Again, as mentioned before this project seems to be very simple when compared to the project at hand. The project discussed does not have any type of microcontrollers or power management system. It also does not have a battery, which it is an essential part of this project. The last part where both project differs are audio effects. While this project has a DSP effects module, the project discussed does not. Even though there are many differences between both projects, it is worth mentioning this project during the research section because it can be used as a learning tool.

Figure 2.1.1.1: Solar Powered Amplifier Project. Used with permission from Instructables.com

The student working on this project has step-by-step instructions on how to create and mount this amplifier into a project box. It also includes specific parts and schematics on the amplifier circuit. All these instructions come as a great benefit to the team as no one in the team had a lot of experience mounting parts to a project box or dealing with solar panels. From figure 2.1.1.2, it can be seen how simple and rustic this project is. At the same time, it was a great stepping-stone for the team to get familiarized with solar panels and project box mounting.

The second similar project that will be discussed is the Solar Amp. This project was the first one to come up in the research phase that actually uses solar panels to charge a battery. This project is made up of a lead acid gel cell 7.2Ah, a premade car amplifier, and two effects pedals. The amplifier on this project provides up to 150 Watts RMS, it also provides two amplification channels into 4 Ohms speakers. It is claimed by the creator that it takes 3-4 hours of charge in order to get about 1 hour of music playing. This amplifier also comes with two audio effects that are created through guitar pedals. The two guitar pedals are powered by the battery and diodes are used to lower the 12 V battery input to the 9 V required by the pedals to operate. This amplifier also includes 3 panel meters, one for charging current, one for battery voltage, and one for consumption current. Below is a picture of the Solar Amp both from the solar panel side and the side showing the connections of the pedals and the amplifier.

Figure 2.1.1.2: Solar Amp. Used with permission from Jason Waddell

From both of the figures above it can be seen that this project is much more sophisticated that the first project described in this section. The project casing is sturdier and the amplifier actually includes some audio effects. It can be said that out of all the researched solar powered amp projects, this is the one closest to the team's senior design project. There are some major differences that are worth mentioning in this case. Everything used to create this project is pre-made. For the senior design project, every part of the project was designed by one of the team members, which adds a level of difficulty that was not present when this project was built. As it was mentioned before both the amplifier and the pedals were bought and not designed by the creator of this project. This project also lacks on the display department. Very few of the researched amplifiers actually came with an LCD. For this project, it is not strange that it does not come with a display since it also does not have a power management system and there would not be any relevant information to display on the LCD. It is claimed by the project's creator that the amplifier works very well. Overall, the Solar Amp has some relevance when related to the project described in this paper. The areas where this project can directly impact the senior design project is the solar panel mounting and charging circuit and the project casing. The creator of this project has a lengthy discussion on how to encase the amplifier, which can be of great help to the team as the project box is one of the areas that are not completely decided. The author also provides the schematic of the circuit that charges the battery using the solar panels. This schematic is also helpful when determining the battery charging circuit.

2.1.2 Commercial Amplifier Products

It came as a surprise to the group that there are no any products in the market that closely match the description of this project. During the product research, the team found that a lot of the commercial products partially matched this product but not completely. The hardest feature to find in commercial grade products was solar panels. The products described below are the ones that most closely matched the product, but as mentioned before, there was not one product that was identical to this project.

The first product discussed will be Roland Mobile Cube. This amplifier is a battery-powered stereo amplifier. It supports several inputs, as guitar, keyboards, computer audio, and MP3s. It also includes built-in audio effects as overdrive, chorus, delay and reverb. The effects are controlled through potentiometers. One of the reasons that make this amplifier so interesting is the fact that it runs on batteries so it is completely portable. It is also very small. Only measuring 11-1/16" (W) x 4-1/4" (D) x 7" (H), and weighting 5 lb. 9 oz. It also provides about 5W stereo power. This product retails for about $169.00.

This amplifier includes a lot of the qualities that were desired for the amplifier in this project. The size that this amplifier presents is excellent for a portable device. This amplifier also includes many of the effects that were desired for this project. Given how small the amplifier is, the effects are most likely created through a DSP chip, which is desired for the team's amplifier. Overall this product seems stay close to some of the project's desired specifications. Some other specifications are not closely matched by this product. One place where the Roland Mobile Cube differs from this project is stereo power. This amplifier gives about 5W of stereo power while the amplifier in this project is designed to provide about 45-55W of stereo power. They also differ on an LCD display. As discussed before, the amplifier in this project includes an LCD display to show battery power. The discussed product does not have any type of display. The last, and most important, difference is the way this product is charged. It was discussed before that this product works through batteries, and the *Unplugged* charges through solar panels. Given that there is not any charging of batteries going on, the discussed project does not have a need of power management system, which were essential when this project was built. Overall, the Roland Cube offers some good features for the price, but the amplifier discussed in this paper will definitely posses more features and some improvements over the ones available with the Roland Cube.

The second product discussed in this section will be the Regen ReVerb Solar-Powered iPod Dock. This product is one of the only solar powered amplifiers found on the market. The first feature to point out on this amplifier is that it only designed to amplify an iPod or iPhone signal. It also stands about 3 ft tall, and is able to provide about 60 Watts of power. This amplifier is also designed to be

charged through a regular wall outlet if solar charge is not desired. It is claimed that the ReVerb takes about 6 hours of direct sunlight or 14 hours of direct sunlight indoors to fully charge, and on this charge, it can play about 12 hours of music at normal volume. The ReVerb offers a 4.5 x1.5 inches backlit LCD. This product costs about $2,229.00.

Figure 2.1.2.1: ReVerb Solar Powered iPod Dock. Used with permission from regenliving.com

Again, as the Roland Cube, the Reverb does not fully match this project's description, but it does come closer to matching what the goals for this project were. The first thing to point out about this project is price. It will be a very expensive amplifier, especially when it is considered that it can only be used with iPods. The second thing to pint out is that it not really portable. Even though the ReVerb is charged with the sun, its size does not really allow it to be a very portable device. The amplifier described in this project had portability as one of its goals. The ReVerb is able to provide a little more watt power than what this project's stereo power. The ReVerb will be able to provide about 5-10 more watts, which should not really make that big of a difference since volume is not linearly related to the number of watts. The ReVerb includes a backlit LCD display that shows how many hours of charge are left on the battery, the level of sun it is receiving, and VU meters. This LCD provides more functionality than what is provided by the *Unplugged*. The ReVerb is pictured on figure 2.1.2 figure 1. Overall, this amplifier, even though different in many ways to this project's specifications, seems to be the only product on the market that most closely fits this project's description.

From the research in the existing product area, the biggest surprise discovered was the fact that there is not one exact product that matches the functionality and specifications of this project. It also came as a surprise that it was hard to find senior design projects that related to the project described in this paper. The fact that not many information on existing products or project is found might be a disadvantage for the team. Information, especially from previous senior design projects, is always desired as any mistakes or problems are clearly stipulated on any documentation found in this area. The previous projects found presented a foundation for the group to start the project on but did not clearly describe a design process or any problems met along the way. Even though, the scarce information on this type of products raised some concerns for the group, it also pointed out that this project could be a marketable project. On section 2.1.2, the Reverb was discussed along with its elevated price of $2,229.00. This price is more than what the stipulated budget for this project is. After examining, and researching the Regen ReVerb, and considering its hefty price for being just an iPod amplifier, it became clear that the amplifier described on this project was well budgeted for. The amplifier that the team built not only amplifies an iPod input, it also amplifies a microphone or a guitar input. Considering all this, the idea of marketing this project becomes more appealing, especially due to the fact that there is not any product in the market that meets this characteristic. It was assumed from the beginning of this project that many of this type of product would be found on the Internet. The research in the area of commercial products and existing products was invaluable as it opened doors into the future of this project and created a drive for the members of the team to achieve a project, that once designed and built, could be the first of its kind.

2.2 Solar Panels

Solar panels are made of many photovoltaic cells connected in series or parallel. These photovoltaic cells are made of different semiconductor materials like silicon. These materials are used because of the electrons they have in the valance gap. This electrons produce energy once light is emitted to the solar cells. Electric field makes electrons move freely which leads to the flow of current in one direction. This and the voltage of solar cells are the parameters used to get the watts that solar cells produce [1].

Using solar cells has many advantages. First, there are no moving parts. They also have long life and are portable, which are some of the reasons to use them for power systems. The highest efficiency for a commercial solar panel is 19%, which means that it is going to convert 19% of the solar light emitted received into electricity. These solar panels are manufactured by SunPower Silicon Valley, but in general other companies have said that they have reached efficiency of

less than 30% [2]. Different types of solar cells exist such as thin films, single crystalline silicon, and polycrystalline among others.

2.2.1 Crystalline Silicon PVs

There are different types of crystalline solar cells. The first one is the single crystal silicon cell, which are expensive to produce as they need to be grown using the Czochralski method. On the other side, multicrystalline silicon PV cells are less expensive to produce, but less effective. These cells are produced by melting silicon, which is going to produce rectangular ingot of multicrystalline silicon that is cut into blocks that are then put into thin wafers [3]. Overall, crystalline silicon is considered to be a poor absorber of light reaching a maximum cell efficiency of only 24.7% for the single crystalline silicon and 19% for multicrystalline. To produce an output voltage to charge a 12V battery, 36 cells of crystalline silicon are soldered together in series [4]. These cells are protected against weather effects and have warranty of 25 years.

2.2.2 Thin film PVs

Thin film panels are produced by depositing thin layers of photovoltaic materials on other materials such as plastic, stainless steel or glass [4]. Then, by using laser technology single cells are created.

Some thin film technologies have reached efficiencies above 13%. The cost of producing thin films cells is less expensive because all the cells are together; therefore there is no need for them to be soldered together and mounted in frames.

2.2.3 Amorphous Silicon PVs

Amorphous silicon is the most common and developed thin film technology. Solar panels made of this material can continue to charge at cloudy conditions. On the other side, one clear disadvantage of amorphous silicon PVs is their short lifetime because their power output decreases by 15-35% with the direct exposure of the cells to the sun. Also, their efficiency ranges from 6-8%, which is smaller than that of crystalline silicon [5].
Furthermore, the production of this silicon material is more complex because it is mixed with germanium to improve the light absorption by reducing its band gap [4].

Table 2.2.1 below shows maximum values for different types of solar cell materials.

2.3 Batteries

Nowadays, the use of rechargeable batteries has been critical in the design of new technologies. There are many different rechargeable batteries types, but the scope of this research will cover the four most important and widely used types.

2.3.1 Lithium-ion Batteries

One of the most important characteristics that lithium-ion batteries offer over other batteries is that they have high energy densities, which means that they can store more energy per space and weight than other batteries [9]. Lithium-ion batteries are widely used in portable devices. It has an energy density which doubles that of Nickel Cadmium. Also, they have no memory so there is no need to discharge the batteries completely for them to give a 100% and the self discharge rate is lower than that of other batteries, which explains why it retains their charge for longer times.

This type of rechargeable batteries uses constant voltage to recharge them. This process of recharging is completed once the current drops below the current limit, which is set by the manufacturer. It also uses a current limiter to protect the battery from overheating [9]. One clear drawback of using lithium ion batteries is that they need to have a protection circuitry in order for them to work safely. What this does is that it limits the peak voltage of each cell to prevent them from reaching a low voltage during the discharge process [11].

They also have a higher cost of production than NiCad, which is around 40% less expensive. Li-Ion batteries produce the same energy as Nickel Metal Hydride batteries but have a weigh which is 35% less than NiMH. This is beneficial for portable devices such as laptops, cameras, cell phones, among others because battery weight is very important when deciding which portable to buy. Nowadays portability is very important. The disposal of such batteries is less harmful for the environment because they don't contain toxic materials such as Cadmium or Mercury [8]

2.3.2 Sealed Lead Acid

Sealed Lead Acid batteries, also known as gel-cell batteries are also charged by using constant voltage. A big difference between SLA and Li-I batteries is that SLA are less expensive. They are usually used for alarm systems and also use a current limiter to protect them from burning. One advantage that it offers is that it can be charged for long times as long as the cell voltage does not exceed the specifications [9]

A lead-acid battery is an electrical storage device that uses a reversible chemical reaction to store energy. It uses a combination of lead plates or grids and an electrolyte consisting of a diluted sulfuric acid to convert electrical energy into potential chemical energy and back again. The electrolyte of lead-acid batteries is hazardous to your health and may produce burns and other permanent damage if you come into contact with it [7].

This battery is used when very high power is required and is the most powerful type of rechargeable battery on the market [8]. This type of batteries performs well at high temperatures as they have been used on cars. Sealed Lead Acid battery weighs more than other rechargeable batteries. They weigh twice as much as the rest of the other rechargeable batteries and if this battery is completely discharged it can be damaged with acid and will no longer work. There is no way to fix this type of battery after this happens. For this reason this battery needs to be constantly charged to expand its life time.

2.3.3 Nickel Cadmium

Nickel Cadmium recharging process is different from other batteries. NiCad uses constant current to recharge itself. It is stated that it can be recharged over a 1000 times, but suffers from a high discharge rate [9].

The Nickel Cadmium battery was created in 1899, but began to be produced in the United State in 1946. It is an old technology that has been in production for over 50 years. One advantage of NiCad battery is that it has a long charge and discharge cycle, which means that this type of battery does not recharge quickly and discharges after long time. Nickel Cadmium batteries have the lowest discharging rate of all rechargeable batteries excluding Sealed Lead Acid batteries. They have a relative high current which is the principal reason for them to be used in cordless power tool. This battery has an average life of 1000-1500 charges.

Compared to Nickel-Metal Hydride cells, NiCad batteries can retain energy for longer times. One big problem that Nickel Cadmium suffers is that it creates memory, which happens when the battery is recharged when the battery is not completely discharged. Moreover, if the NiCad battery is charged after it reaches it maximum charge value the amount of storage could be reduced because as it is stated in [8], if the user let that happen the contact crystals in the battery are going to increase in size; hence, the battery will have less area for energy to be stored.

2.3.4 Nickel Metal Hydride

Nickel Metal Hydride batteries are known to have a higher energy density than NiCad, but using them have more disadvantages [9]. These types of batteries are widely used in portable devices such as notebooks. One of the drawbacks of these batteries is that they need to be recharged carefully because if they are overcharged they can be damaged greatly. Another disadvantage that Nickel Metal Hydride batteries have over other types of batteries is that they have a discharging rate of 20% per month.

Nickel Metal Hydride batteries have a capacity to store 30% more energy than Nickel Cadmium ones, but their charge cycle is less. These types of batteries do not create memory if they are discharged completely, and recharge faster than Nickel Cadmium batteries. Different from Lithium-Ion batteries the disposal of such batteries is not harmful at all for the environment.
As stated before, these batteries are widely used in notebooks. It is important to know that new batteries do not charge as they are supposed to do when they are new. Because of this, it is recommended to charge the battery completely and use the computer until the battery dies, by doing so the battery will reach its full capacity. This process is called the cycling the battery [8].

2.4 Solar Power Management Microcontroller

The microcontroller is a very important part of the project. It is what powers the whole system and makes it smart. In order to find an efficient yet simple to program microcontroller it was important to make some considerations like the amounts of outputs and inputs needed for the whole connection of the solar panels with the circuits, the A/D converter and the battery circuit.

2.4.1 Arduino Mega 2560

The Arduino Mega 2560 was the first microcontroller taken in consideration. Arduino boards are well known because they are an open-source hardware computing platform. It is a simple input/output board that can be programmed using C language by using an open-source IDE. This microcontroller has 32 registers that are directly connected to an Arithmetic Logic Unit [13].

This microcontroller has:
4 Kbytes EEPROM
8 Kbytes SRAM
54 digital input/output pins
14 PWM
16 Analog inputs
4 UARTS

1 16 MHz crystal oscillator
256 KHz flash memory
USB connection

Aside from this, the microcontroller can be powered with AC to DC adapter, a battery or by plugging it to a computer via the USB port. It also has an input voltage of 7-12V and needs 5V to be powered.

2.4.2 Texas Instrument TMS320F2808

Another microcontroller taken in consideration is the Texas Instruments TMS320F2808. This DSP microcontroller has 32 bits and has a speed of 100MHz, which is 5x greater than the Atmel mega microcontroller. It also includes a 12 bit A/D converter that has a conversion speed of 160nSec. Other characteristics are shown in table 2.4.1. The advantage of this is that there will be no need to use an external A/D converter and everything will be together in one piece, which makes the design more optimized an ideal for renewable
energy applications [10]. Also, it can be programmed using code composer studio with C/Assembly.

TMS320F2808 Characteristics
Low power consumption 3.3-V I/O
64K x 16 Flash memory
18K x 16 SARAM memory
16 PWM channels
32-bit core
150 MHz conversion for 12-bit ADC

Table 2.4.1: Shows the different characteristics of the TMS320F2808 MU.

2.4.3 Atmel AVR 8-Bit Microcontroller

The Atmel AVR microcontroller is another microcontroller that includes an A/D converter on same chip. It has EEPROM, and flash memory. It can be reprogrammed using fast In-System Programming (ISP) after it is assembled [9]. It can be programmed completely using C, which makes it simpler to program and modify compared to other microcontrollers that need to be programmed using assembly.

This microcontroller is good for battery chargers as the EEPROM data memory can be used to store battery characteristics, such as charging processes. The 10-bit A/D converter allows batteries to reach their maximum charge value. With this, the design will be also optimized and the cost will be reduced.

2.5 Power System Topology

There are two topologies taken in consideration for the design of the power system: path selection and direct topology. As stated in [6], the purpose of using a direct connection topology is to isolate the photovoltaic cell or any other power supply from the battery and system by connecting the battery pack positive terminal and the charger stage output to the system power bus, as shown in Figure 2.5.1. In such a system, the maximum power delivered from the PV panel to the system power bus is limited by the charger settings; the external supply is isolated from the system power bus by the charger power stage.

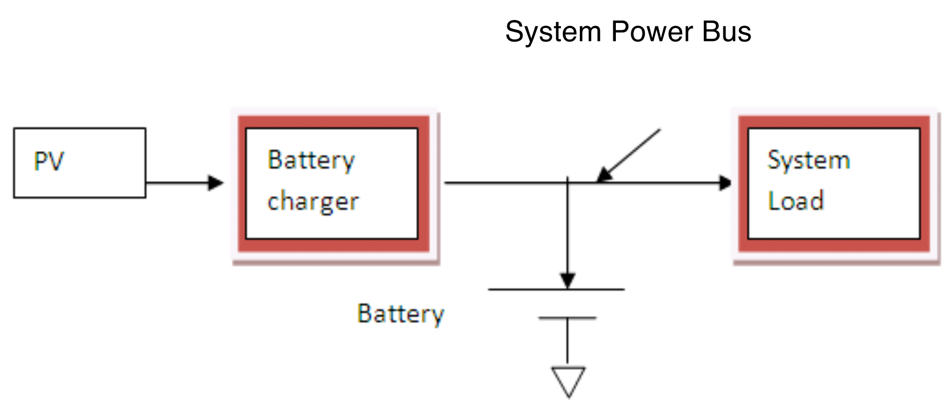


Figure 2.5.1 Direct Connection Topology

Using Direction connection topology has its pros and cons. As stated above, it is used to isolate the photovoltaic cell or any other power supply from the battery and system which have the following advantages:

1. The first advantage of constructing a direct connection topology is that it is easier to implement and less expensive than path selection topology because it does not need switching networks like MOSFETs and BJTs to divide the power.
2. Another advantage is that the total charge current and the system current can be limited by setting the charge current to a value chosen [6].

Also, there are various disadvantages of using direct topology on power systems.

1. As it is stated by TI's Implementation of Battery Charger, if a system requires high current the charging process will never end; hence, the battery is going to be always charging which reduces the life time of it.
2. Also, the battery may not charge completely and if the battery is low or completely depleted the system voltage is going to be cut and will not work.

On the other side, in path selection topologies, the input power is divided between the charger stage and the system load. This topology is shown in Figure 2.5.2. As it is shown in the figure the PV panel is directly connected to the system power bus and the power is divided by using a switch.

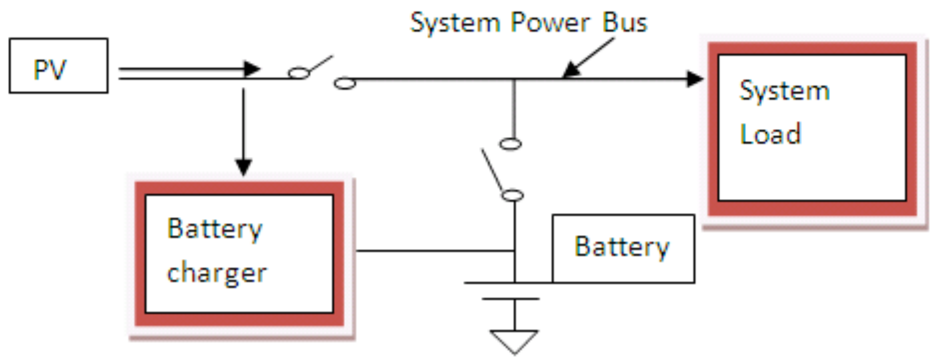


Figure 2.5.2 Path Selection Topology

Using this topology has various advantages.

1. The first notable advantage is that the battery charge is not affected by the load of the system, this happens because the battery and the loads are independent of each other as shown above in figure 2.5.2.
2. It also is able to make the power system reach high load current.
3. The most important advantage over the direct topology is that the system will still work with low or depleted batteries because the PV panel will provide energy required for the system to power up.
4. Also the efficiency can be higher than that of direct connection topology because there will be high voltage difference between the input and system voltages [6]

This topology also encounters different disadvantages.

1. Different from the direct topology this topology requires the use of switching networks, which makes the cost of building power systems with this topology to be more expensive and difficult to build.
2. There is more variation in the system voltage.

2.6 DSP Component Research

The advantages to choosing a DSP component over an analog component for the audio processing that was implemented with this project were vast. Since digital audio processing operates mathematically on a binary representation of

the signal, this allows a design of many audio effects through a small piece of hardware which, in change, will help to meet the size requirements that were set for this amplifier. The only concern that choosing digital signal processing brought was the fact that it was not able to provide perfect filtering, demodulation and other functions because of mathematical limitations. This can make the chip prone to some loss of signal. After thoroughly researching available options and with low power consumption as one of the main concerns for this project, the conclusion was reached that the techniques of digital signal processing are much more powerful and efficient than analog domain signal processing and they fit the requirements for this project.

As research has advanced, there were many DSP products that were found fit for this project. The main goal with a DSP component was to find a good quality product that met the budget, had low power consumption, and also that could be programmed very quickly as there was very limited time to implement our project.

2.6.1 V1000SP Chip

The first style of DSP chip that was considered for potential use in this project was the V1000 Digital Multi-Effects DSP. The V1000 is a DSP chip that has 16 built-in reverb and multi-effects, and it also very affordable as each chip runs for about ten dollars. This chip only needs a 4-bit microcontroller in order to implement the effects that are already built in.

One of the biggest advantages to choosing this chip was the fact that many effects that are desired for our project were already implemented. For instance, echo, phaser, chorus, and flanger came already implemented. Added to the implemented effects on the chip, is with a serially programmable SRAM. This SRAM is used for either program development or dynamically changing programs in case other effects need to be implemented. One of the disadvantages of choosing this chip was that fact that a 4-bit microcontroller was be needed. By having to add one extra microcontroller to the project, the budget could have been exceeded. An extra microcontroller meant higher power consumption than it was originally planned. The project complexity also incremented, and the ever-present risk of not meeting the deadline was increased. Another disadvantage to this chip was that it is not widely used. When compared to other DSP components that have been researched, not a lot of information was found on this chip, other than a couple blogs and the datasheet. The goal for this project was to pick components that are widely used, as by doing this, it was guaranteed that the there was availability of information on how

to use the component. It could be said that this was almost as important as any of the other factors that were considered when choosing a DSP component. Finding information that is readily available on working with the chip could shorten programming time and also simplify implementation.

2.6.2 Arduino UNO Microcontroller

Another alternative for a DSP component considered for the implementation of the project was the use of Arduino Uno Board. In this case the Arduino Uno Board would be used for real time audio processing. The Arduino Uno is a microcontroller board based on the Atmega328 microcontroller.

The use of this board for audio processing had many advantages that related to the project's implementation. One of the first and most important advantages was the fact that these types of microcontrollers are widely known and used. As it was mentioned earlier, one of the most important criteria when picking a DSP component was to easily find information on how to program and use the component. This requirement was fully met with this microcontroller. Information and previous projects using this component are vastly found by a simple Internet search. Another advantage to choosing this component was that it is an open source product. By being open source, the price is lower compared to other microcontrollers that were researched. This low price would help to stay within budget, which as mentioned above, was also one of the main requirements. As this component is a microcontroller itself, the need for a 4-bit microcontroller that was discussed on the previous component option was eliminated. One of the biggest disadvantages to choosing this component was that, as it is a microcontroller and not a DSP chip, there was a need to add some extra components to be able to implement the sound effects that were desired for this project. This included a potentiometer to control the effects, capacitors, resistors, and some type of filter on the audio output. This complicated implementation and design since exact component values had to be chosen to be able to create the effects that were desired. All these new components also consumed extra power that was not accounted for, and created the risk of not meeting the deadline if picking the exact components took longer than the estimated time.

2.6.3 ADAU1702 Chip

The last option discussed for the DSP component is the ADAU1702. The ADAU1702 is a single chip audio system that includes audio DSP, ADCs, and

DACs. This chip was chosen as one of the options for several reasons. The main reason was that this chip already includes AD/DA converters. This saved some space on the PCB as well as making cable connection simpler. Another major advantage to this board was the way it is programmed. Analog Devices provides a program called "Sigma Studio" which provides an easy to use graphical user interface. This GUI allowed for the programming of the DSP chip through graphical components instead of the usual DSP programming needed on other chips. This signified a faster programming time that what was originally estimated. Considering the time constraints that were set for this project, this type of advantage was very important. One of the main concerns when choosing a DSP component was the way that the audio effects were controlled. Fortunately, the ADU1702 chip has an on chip EEPROM, which could potentially avoid the need for a microcontroller to control the audio effects. The EEPROM is controlled through push buttons. This was an ideal alternative for our project. By avoiding a microcontroller for the DSP chip, it helped to stay within budget and also minimize power consumption. This covered two of the main concerns when choosing a DSP component. Considering that this chip could use an 8-bit microcontroller, the option of using one microcontroller to control both the DSP chip and the LCD display was also researched. This option was explored fully on other sections.

One disadvantage of choosing this chip was price. Both this chip and evaluation board were more expensive that previous options discussed above. This was a disadvantage that was not taken lightly as staying on budget was one of the main concerns in this project.

When comparing the previously discussed options, it was found that they offer advantages and disadvantages that were widely different. The first option discussed, the V1000 Digital Multi-Effects DSP offered little programming but it required a microcontroller that was not budgeted for. The second option had an embedded microcontroller but was not a DSP chip, so to do real time audio processing extra components were needed. The third option seemed to have the most advantages that related to the goals for this project, but there was a need to work around the budget since it was more expensive that what was initially considered.

After all the research for DSP components and due to the many advantages that the ADU1702 DSP chip offered, it was the most likely be implemented in the design. Even though this chip was more expensive than what it was initially

budgeted for, this issue was easily resolved by avoiding the choice of a microcontroller by using the on-chip EEPROM. It is believed that that this chip would allow to create our audio effects faster than it would take with any of the other two discussed options. The AD/DC converters were also of interest as this could save space on the PCB.

2.7 Microcontroller for Display Research

There are several concerns that were taken into consideration when choosing a microcontroller component for the project's display. The first, and most important, consideration was power consumption. The device chosen for this project must consume very low power. The lowest power consumed, the better since the project uses solar power to charge a battery. Once this battery is fully charged, it is solely responsible for powering the device. By trying to keep power consumption to a minimum, the charge on the battery lasts longer which was one or our main goals. The second consideration when choosing this component was budgeting. It had to be a device whose price fits into the budget for this project. The last consideration was that it could adequately control a display. The main task of this microcontroller was to control a display. This later consideration was weighted heavily when choosing a device.

2.7.1 Microchip PIC18F87J90 microcontroller

The PIC18F87J90 has intergraded display drive capabilities up to 192 pixels. It is also an 8-bit microcontroller, which was an advantage for the project when it was considered that the DSP component chosen could be controlled with an 8-bit microcontroller. Not only could this microcontroller be used for the display but it could also be used to control our DSP component. This was a great advantage of this microcontroller option. Another great feature that this chip offers was the nanoWatt Technology, which can reduce power consumption by using alternate run modes, multiple idle modes and on-the-fly mode switching. This microchip microcontroller operates at an input voltage range of 2 to 3.6 volts, resulting in a relatively low power microcontroller. From figure 2.7.1 it can be observed that the PIC18F87J90 microcontroller provides 64-128 KB of flash memory, 2 comparators and an LCD Driver, which will not only met, but exceeded, the requirements for this microcontroller to work in our project.

This Microcontroller seemed to be a good fit for our project. It met our budget and power requirements. It also fits our display interfacing consideration. One more

benefit that is worth mentioning for this device, was that microchip provides a segmented display designer GUI which allows programmer to easily create segmented display code by dragging and dropping elements onto a screen. This was a great benefit considering the time constraints of the project and the fact that no person in the group had programmed a display with a microcontroller before. Currently from the researching phase, this microcontroller does not have any major disadvantages specifically related this project.

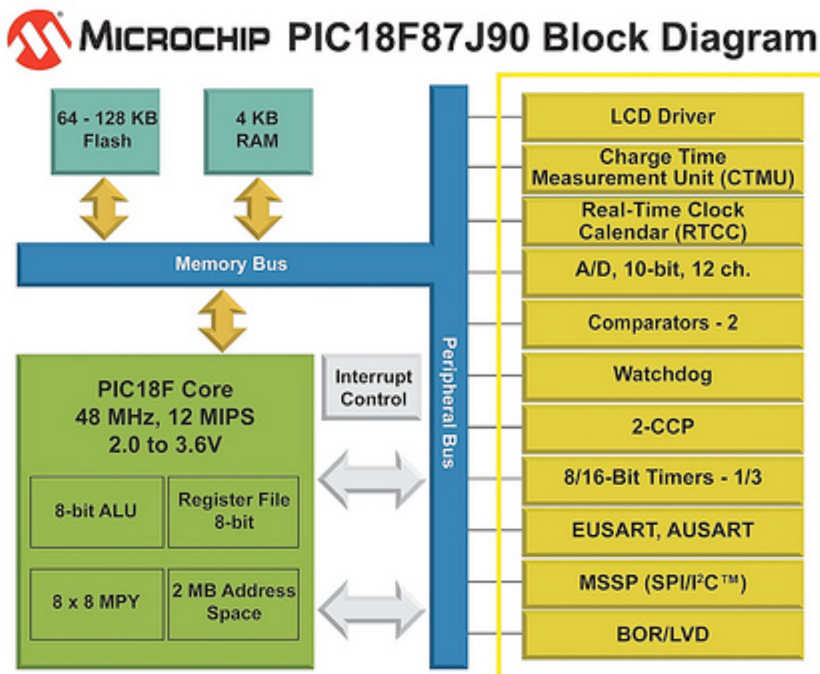


Figure 2.7.1: PIC 18F8790J90 Block Diagram. Used with permission from Microchip.com

2.7.2 Freescale MC9S08LH64 microcontroller

The MC9S08LH64 microcontroller features a very low power consumption operating at input voltage range of 1.8 to 3.6 volts. It is considered best in class standby power consumption, which was a great benefit when considering the goal of low power consumption that was established for this project. This microcontroller provides up to 288 segments for the display. The LCD drive included with this microcontroller can drive either a 3V or 5V LCD glass, which provided flexibility when selecting the display component used for this project.

The MC9S08LH64 also includes two 32KB flash arrays, which provided enough space for the code.

On fist glance, the MC9S08LH64 microcontroller appeared to be a good choice when selecting a microcontroller. It met most projects expectations and goals. Its low power consumption was backed by the Energy Efficient Solutions mark, and its display capabilities seemed to exceed expectations. On further review of this microcontroller, it was observed that it provided more functionality than what the project required. The display drive offered more segments that what was needed to implement the display. The microcontroller also offered functionalities that can be applied to the medical field, and that will evidently was not used in this project. After examining this option, it was clear that even though this chip met the project's goals, a simpler microcontroller could be used to achieve this project's required functionality.

2.7.3 Arduino Pro

The Arduino Pro microcontroller was also a great choice for this project. It comes preassembled with the ATmega168 or ATmega328 chip depending on the model. It also comes with 16 KB of flash memory, 14 digital input/output pins, and either a 8 MHz or 16 MHz clock. It also offers the choice of two operating voltages, 3.3V or 5V. As low power consumption was one of the main goals of this project, the 3.3V operating voltage would definitely be chosen over the 5V version since it will draw less power and last longer on battery power. The input voltage range for this board is 3.35V to 12V, which was somewhat high compared to the options discussed above. One big advantage of the Arduino Pro was that it can be programmed using a C compiler, which could greatly help to meet the deadline since C is a language known by all the members of the group.

The Arduino Pro offered many advantages that impacted this project directly. All of Arduino boards are open source products. By being open source, the price not only met the budget requirements, it surpassed them. Another advantage to being open source was that sample code was readily available on the Internet helping the members of the team to get familiarized with programming the Arduino board. There are two main disadvantages that were found from the research on this board. The first was the fact that this board consumed more power than the previously discussed options, and the second was the fact that this board does not have a dedicated display driver as the other options did.

Even though the Arduino had some great features; it also had some major downfalls when compared against the projects goals.

2.8 LCD Component Research

There were two main goals that needed to be met when selecting an LCD for this project. First, and most important, the display had to be compatible with the microcontrollers that were mentioned above on section 2.7. The display also needed to provide enough functionality so that the required information from this project could be displayed. Another important goal that the LCD needed to meet was low power consumption. As it was constantly mentioned throughout this project, low power consumption was a goal that all of our components had to meet so that battery life was maximized. The last goal for this display was to meet the budget requirement.

2.8.1 Hitachi HD44780 LCD

The HD44780 was the first option that will be considered to use in this project. Some of this LCD features are:

- 5 × 8 and 5 × 10 dot matrix possible
- Low power operation support: 2.7 to 5.5V
- Wide range of liquid crystal display driver power 3.0 to 11V
- 4-bit or 8-bit MPU interface enabled
- 80 × 8-bit display RAM (80 characters max.)
- 9,920-bit character generator ROM for a total of 240 character fonts

The first advantage that was noticeably good for this project was the low power consumption. This would allow staying within the power budget, a main issue in our project. Another great feature of this display was price. This screen can be bought starting at $7 on a 16x2 size. This price is a lot less than what the display had been budgeted for, so it was a great fit for this project. It can also be observed from the list of features that it works on either a 4-bit or an 8-bit interface, which provided flexibility when the programming is performed. The HD44780 is also a widely used display. Information on how to program this display is readily available, and a C compiler can be used, which was an added advantage since no member on the team has programmed an LCD before. Since the main information displayed on this LCD is battery life, it seemed to be a great fit for the project needs.

2.8.2 Nokia 3310

The second option discussed for a display is the Nokia 3310. Some of its features are:

- Voltage: 3 - 3.3 V
- Dimensions of 40 x 38 mm
- 2 modes of display, normal/inverse
- 84 x 48 pixels

From the list of features, it can be observed that this LCD would also meet the project's power budget, as the previously discussed option. It would also meet the monetary budget, as it runs for about $6 on the Internet. Even though it meets some goals for this project, this LCD falls short on certain aspects when compared to the other LCD choice for this project. One of the main aspects where this LCD differs from the previous choice is availability of part and existing information. The Nokia 3310 is not as popular as the HD44780; therefore, existing information on the part is hard to find, and previous projects using the part are very limited. This fact impacts the project greatly as previous projects and existing code on the Internet are ways to get familiarized with how to program the LCD. Finally, This LCD display is widely used for phone applications which means that when the part is bought from a vendor on the Internet, they will usually sell the LCD to be interfaced with the phone. Because the LCD would not be used in this way, then it would require a PCB board to be created. This would cost the group money and precious time that could be used on a most important activity. After much research of this LCD, it is concluded that its disadvantages would not make it the best choice to use in this project. Figure 2.8.2.1 displays a picture of the Nokia 3310 when used for a thermometer application.

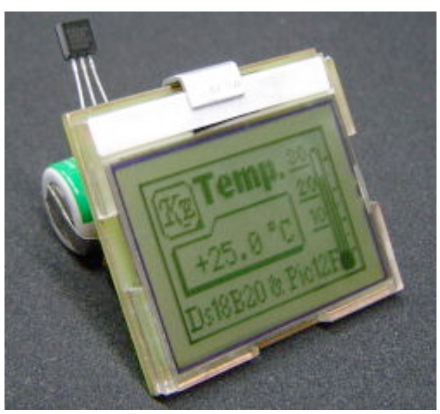


Figure 2.8.2.1 : Nokia 3310 Thermometer application. Used with permission from neggadget.com.

2.9 Audio Amplifier Technologies

2.9.1 Valves

Before the inception of the BJT transistor in the early 1950's the only active electronic devices where vacuum tubes. The basic principle of operation of the vacuum tube was thermionic emission. This is when a relatively high voltage thermally excites a cathode and it discharges electrons into a vacuum. These devices that could be used as signal amplifiers or switches. Valves are very attractive to audiophiles and audio equipment designers because of their linearity. Linearity in this context is their ability to take an input with a complex waveform and multiply each and every one of the frequency components of that signal by a constant factor. This goes back to the original principle of an amplifier that does not favor any given frequency of an input and is able to take in a small audio signal (20 Hz-20 kHz in the 100's mV range) and output the same exact signal with the same frequency content but at a much larger voltage. Also when they are overdriven their distortion has a particular clipping characteristic in which when the transient of an input waveform is "chopped off" it is done so gradually instead of abruptly (like solid-state devices). This is the one and foremost characteristic that still drives audiophiles to go through the trouble of localizing valves and valve amplifiers.

The main disadvantages of valves are that they are very difficult to manufacture, are extremely fragile and their power consumption is very high in comparison with modern solid-state components. Valves are bulky when comparing them to modern transistors, they posses an anode and a cathode encapsulated in a glass container that needs to have as much air evacuated from it as possible. This is a very expensive and cumbersome process. The fact that the casing of the Valve is made out of glass adds to the fragility of it. A careless bump on the device or even in the amplifiers case would ruin the valve. Lastly valves are power hungry devices. In order for valves to work they must possess specialized power supplies (adding to the bulkiness) able to deliver anywhere from 200 to 500 Volts and above. This makes valve amplifiers potentially dangerous to people unfamiliar with their operation principles. Having a power supply that steps up voltages this high adds expensive transformers to the power supply as well as the potential trouble of mains voltage filtering into the audio amplification circuitry, this bit about mains voltage is true for all amplifiers, but it is specially troublesome for such high voltage devices.

In short, valve amplifiers were the first solution for the need of amplifying and audio signal as linearly as possible. A lot of the principles that make valve amplifier operate are used for modern solid-state based designs. However, their modern use is limited to a very small audiophile community and some older RF frequency amplifiers.

2.9.2 Solid-State Amplifiers

Class A

Electronic amplifiers can take as input either current sources or voltage sources and source out voltages or currents. This translates into 4 possible configurations, which are the current controlled current source, the voltage controlled current source (transconductance amplifier), the current controlled voltage source (transimpedance amplifier) and last but not least the voltage controlled voltage source. Of these four possible combinations the one most relevant to our study of electronic amplifiers is the last one, voltage controlled voltage sources. This is so because all audio signals from either instruments or electro-mechanical transducers (microphones) is transmitted as an AC voltage and is expected to go through several analog signal processing stages (pre-amps, amplifiers, filtering) and then be reproduced through a load (speaker) in the form of a larger AC voltage. Class A amplifiers are the simplest configuration of these electronic amplifiers. They basically consist of active electronic devices (BJT's, FET's or Valves) and modulate the output with respect to the input by taking potential energy (voltage) from the power supply or rails and shaping the input signal into a replica of itself with a larger voltage swing. Class A amplifiers are single-ended systems, which means that one single device takes care of reproducing the amplified output waveform for both the positive and negative cycles of it. Also, the concept of always-on amplification deserves a little further discussion. Audio amplifier designers more accurately refer to this concept as the conduction angle quantity. The conduction angle is analyzed as a circle which means that a 360 degree angle refers to a system that has an input that is "on"throughout the whole time of amplification. In the other hand a 180-degree angle of flow refers to a system that is only amplifying through half of he input cycle. Since the input is a waveform it has a positive cycle and a negative cycle and a class A amplifier is on for the full duration of this cycles it is said that a class A amplifier possesses a 360-degree angle of flow. Older amplifiers for TV sets, transistor radios and guitar amplifiers used class-A topologies.

As is the case with all devices that use BJT transistors, 3 possible combinations are possible: Common-Base, Common-Emitter and Common-Collector or Voltage follower. For the sake of briefness only one of these kinds will be discussed: the Common-Emitter. Like any device that uses NPN transistors careful calculations for the system components must be done to maximize voltage swing and minimize the DC off-set of the output with respect to the input. All of the body of work derived for NPN transistors applies. The assumptions needed to design a working audio class A amplifier are simplified by the fact that the only critical bandwidth of operation is within 20 Hz and 20 kHz. In the other hand careful consideration must be made of decoupling any DC component of the output waveform before delivering it to the load, in this case a speaker. This goes back to the way a speaker works. The diaphragm inside of a loudspeaker moves back and forth in accordance to the variations of current delivered from the audio amplifier. If a current is "pushed" into the speaker (positive polarity) the diaphragm will move forward, if a current is "pulled" from the speaker (negative polarity) the speaker will recede. Speakers are usually calibrated in such a way that when no DC component is present in the signal the diaphragm rests at exactly the middle of the points of minimum and maximum vibration within the speaker driver. If any significant amount of DC current is present in the amplified signals the speaker diaphragm will shift from the mid-point to a point forward within the speaker driver. If the applied amplified audio signal is large enough this can cause the speaker diaphragm to detach from the driver enclosure or even break. To illustrate the simplicity of a single-stage class-A topology and the characteristics of its corresponding output one is shown in figure 2.9.2.1.

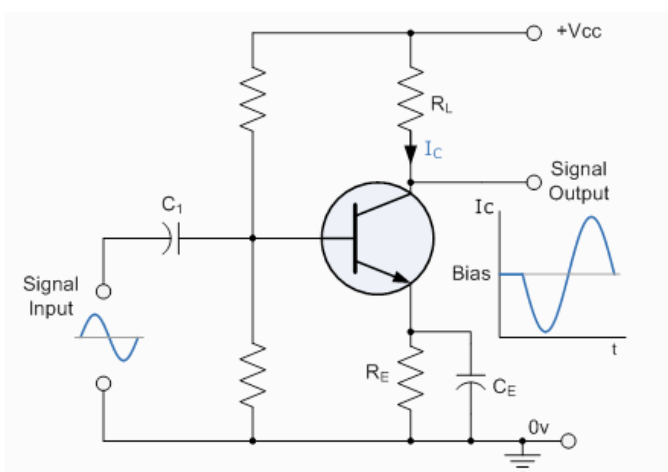


Figure 2.9.2.1: Single Stage Class A amplifier with C-E configuration and corresponding input and output characteristics. Reproduced with permission of Wayne Storr from www.electronics-tutorials.ws

Could this be too good to be true?, a high fidelity reproduction of the small-signal with only one active component, one voltage source and a handful of passive components?. The answer is yes. Despite the ease of implementation of class-A amplifiers they have major drawbacks when it comes to power efficiency. Since the amplifier is being biased and draws a quiescent current constantly it means that the device is radiating power regardless of whether an input is applied or not. Also, when there is a signal applied the single transistor is always on for both the positive and negative cycles of the input wave . A design like this translates in an amplifier model with a relatively low efficiency that can be anywhere from 12% to 25% (Elliot), designs with current mirrors and other extra circuitry but with a class-A output stage are even reported to be up to 40% (Storr). To put this in perspective it means that a single-stage amplifier topology that is designed to deliver 50 Watts must be fed with anywhere from 200W (best case) to 416 W (worst case) of power. The other 82% to 75% of power would be dissipated in the form of heat in the biasing resistors or within the actual body of the transistor. This requires careful and extensive heat sinking of all class-A amplifiers. Despite their horrendous power consumption, class A amplifiers are relatively simple to design. They only need one active output device such as a transistor, and a power supply so they would require less real state on an actual circuit board. Also the active device that takes care of the amplification is the same at the output. This is why class-A amplifiers are usually referred too as single-ended devices. This means that a lot of real state can be saved when building a class-A amplifier, this savings in space must be traded off with the overall lack of efficiency and the need of using big and expensive heat sinks. Like all engineering designs there is always a trade-off. In modern amplifier designs class A amplifications is reserved for certain applications where power efficiency is of little relevance to the operation of the system because their use is more seldom. Systems such as building intercoms, public address systems for buildings that use a 70.7 Volt constant voltage power source and even workbench testing amplifiers come to mind.

Class B

An alternative to the “always-on” or 360 degree conduction angle of Class A is the Class B implementation with a conduction angle of 180 degrees. This kind of amplifier possesses two active output devices that complement each other. For the sake of discussion we will assume that they are BJT’s so one would be a NPN transistor and the other one a PNP transistor. In a basic class B topology the emitter of the NPN would be feeding into the emitter of the PNP. Each one of

these output devices conducts current for half of the sinusoidal cycle of the output waveform, This is why it is said that they have a 180-degree conduction. This translates into a device that takes a small signal, breaks it into its positive sinusoidal components and its negative sinusoidal components, amplifies it and then adds up the amplified waveforms to make a scaled up representation of the small signal input. This greatly increases the power efficiency of the audio amplifier since only one of the devices is on at any given time and it is only required for it to amplify one half of the waveform to the rails of the amplifier. By pure logic the power efficiency of a class B amplifier would be placed anywhere between 25% and 50% efficiency, but designs with more active components and high-gain, high-efficiency transistors have been able to generate power efficiency figures of approximate 70% and have a theoretical efficiency ceiling of 78.53% (Self 137) this is of course with no transmission losses or leakage currents of any device which is never the case in practical applications. Theoretically all amplifier classes could be implemented as battery operated devices. However the large quiescent currents of transistors used as Class A amplifiers would make the use of batteries for amplification of music signals very expensive. In the other hand a transistor amplifier capable of delivering a 70% efficiency would require a much smaller heatsink than a class A device with around 25% efficiency. This would make the implementation of it in portable devices very attractive. This is why most solid-state portable electronics prior to the invention of modern switching amplifiers would favor a class B design approach. They can be found in all kinds of portable electronics such as TV sets and transistor radios. In order to further understand the operation mode of class B amplifiers a representation of the principle by which class B amplifiers operate can be appreciated in figure 2.9.2.2.

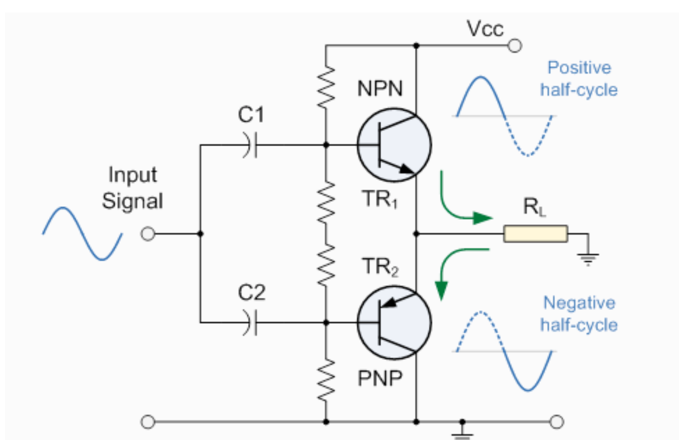


Figure 2.9.2.2: Single Stage Class B amplifier with complementing BJT transistors Reproduced with permission of Wayne Storr from www.electronics-tutorials.ws

The class B topology poses an elegant solution to the problem of audio amplifier power efficiency. Break the wave apart, amplify each cycle by separate and then add them up to get back an original waveform with an increased voltage swing. This approach of generating a positive bias across the base and emitter of the first transistor, hence delivering a current to the load and then proceeding to create a positive bias across the second transistor and creating a current coming from the load is called the push-pull output configuration. The main problem with this kind of topology resides in the PNP transistor in the bottom of the schematic. Since for the transistors to turn-on they must have a 0.7 V bias between the collector and the emitter, otherwise they will not conduct. Theoretically this means that any part of the amplified audio signal that falls below 0.7 V will not be go through the PNP so it will not be added to the final waveform of the amplifier. Likewise, if a bias of 0.7 Volts is not present across the NPN's base and emitter it will not conduct. In reality a certain amount of signal will be generated from all the emitter currents, however it is very small. This will cause for the parts of the amplified signal that fall below the window that exists at the 0.7 volt point and above the -0.7 volt point to not be reproduced with great precision by the amplifier. Since this unpredictable output only occurs when the amplified waveform crosses over from above to below the turn-on voltage of the output transistors it is known as crossover distortion. This is an appropriate point to introduce the concept of AB amplifiers.

Class AB amplifiers

Class AB amplifiers are a compromise within the two aforementioned amplifier classes. They combine a high efficiency push-pull output stage comprised of two complementing BJT's or MOSFET's like the class B amplifier with a scaled down version of the class A approach of "always-on" signals. As far as conduction angle is concerned a class AB design will amplify a waveform anywhere from 182 degrees to just below 360 degrees (Self, 33). In order to understand why one would want to have a conduction angle with such a variable range the concept of crossover distortion deserves further explanation. Crossover distortion must be accounted for in all amplifiers that use a class B push-pull principle of operation. At first glance to figure 2.9.1.2 a reasonable solution would be to introduce some sort of constant forward bias that assures that the transistors are always partly open and conducting. A possible solution would be to include a feed-forward loop within the amplifier design (Self, 222). This loop would consist of an operational amplifier with its output placed at the input signal of the amplifier and taking as an input a copy of the amplifier's output. This kind of configuration would essentially

feed forward a copy of the line level audio signal from an instrument or MP3 player, amplify it by means of the operational amplifier and feed it directly into the amplifiers output. Since typical operational amplifiers like the LM324 and the LM741 have gains in the order of the 100 thousands this means that a low level audio signal in the millivolt range can be increased above the turn-on voltage of the transistor pair, 0.7 Volts in this case, and be fed-forward to the output transistor pair. The main advantage of this is that the transistors will only be biased into forward active mode when an audio signal is played through the operational amplifier located at the input of this arrangement. This operational amplifier could even be powered up with the same rails that power up the push-pull transistor pair and avoid the presence of extra power sources.

A second possible solution to crossover distortion would be to introduce a voltage bias in the form of a couple of reference diodes, once for each transistor. They would be collocated in series and would short the inputs of each complementing BJT's or MOSFETS. This way, they would assure that both devices are always on. The main difference between this always on characteristic of the AB amplifier and the one encountered at the class A design is that using reference voltage diodes introduces a DC voltage bias that is independent of the audio input voltage swing. This assures that each transistor will remain on for the duration of 180 degrees of the waveform plus however many degrees the DC biasing can generate. In essence the bigger the diode biasing of the amplifier the more it approaches class A operation. The main feature of this design is that matter how little the input voltage signal is the crossover gap when one transistor turns off and the other one turns on is reduced proportionally to the amount of biasing. Essentially, the presence of a positive bias in the form of diodes would move the Q point of the transistor higher in the DC load line. This is what reduces the amount of crossover distortion generated by the class AB amplifier and makes it especially attractive for high fidelity audio applications such as home theaters, TV sets, consumer audio amplifiers, car stereos and computer speakers. To further understand the concept of diode pre-biasing and how it affects the push-pull stage of a class AB amplifier figure 2.9.2.2 displays a class AB amplifier with the aforementioned characteristics.

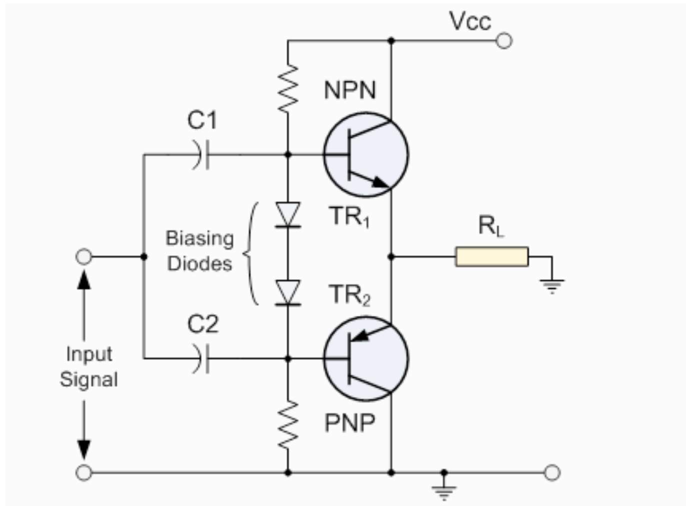


Figure 2.9.2.2: Single Stage Class AB amplifier with complementing BJT transistors in push-pull configuration. Reproduced with permission of Wayne Storr from www.electronics-tutorials.ws

Careful consideration must be made whenever a class AB topology is desired for portable amplifier applications. This is so because the presence of two pre-biasing diodes “steal” voltage from Vcc and reduce the practical amount of quiescent current that can be fed into either of the two complementing transistors from the push-pull stage. Nevertheless the quiescent currents dissipated by modern AB designs are orders of magnitude below the quiescent currents dissipated by class A designs (Self, 35).

Class C amplifiers

Class C amplifiers differ from their A,B and AB counterparts in that their conduction angle is always less than 180 degrees of the input waveform. This means that are extremely efficient and can deliver higher wattage with the same amount of biasing voltage as one of the other kinds of amplifiers. With appropiate filtering and peripheral circuitry they make excellent RF signal amplifiers (30kHz and 300 GHz range). However they are very non-linear in the audio frequency range of 20 Hz to 20 kHz so they are not used for this application.

2.9.3 Switching Amplifiers

Class D amplifiers

All the previous amplifier models discussed relied either on the conduction angle of the input signal and/or the quality of the switching capabilities of output

devices. Class D amplifiers take a whole different approach to small-signal amplification. They can have either analog or digital signal inputs, we concentrated on the former. Essentially the small-audio signal voltage is fed through an operational amplifier and modulated with a high frequency triangular wave generator. This wave will act as a carrier of the small-signal voltage. Usually this carrier frequency is ten times larger than the highest of the frequencies in the audio signal. This means that frequencies above 200 kHz are ideal for this modulation task, in practice frequencies as high as 400 kHz are used (Self 33). The resulting square wave is then fed through a complementary MOSFET pair that will switch the instantaneous value of the signal between the actual value of the rails. This will amplify both the original audio signal as well as the high frequency carrier which will then be removed by feeding it through a low-pass filter. The most attractive principle of class D amplifiers are that since the amplified output is going back and forth between the rails of the amplifier at a very high frequency, the MOSFETS taking care of the amplification have very little time to turn on and off. This translates into a very cool (as in temperature) and efficient mode of operation. Efficiencies in the range of 90% and more are not unheard of. This also makes class D amplifiers extremely small since they do not require extensive heatsinking. This makes them extremely popular for self powered speakers, DVD players, Modern TV sets and more. Figure 2.11.3.1 offers a visual representation of what exactly happens to the analog voltage input through each one of the stages of the class D amplifier. From a sinusoidal waveform to a square waveform with a relatively low voltage to a much larger square waveform with a magnitude equal to the rails of the amplifier to an amplified copy of the original input.

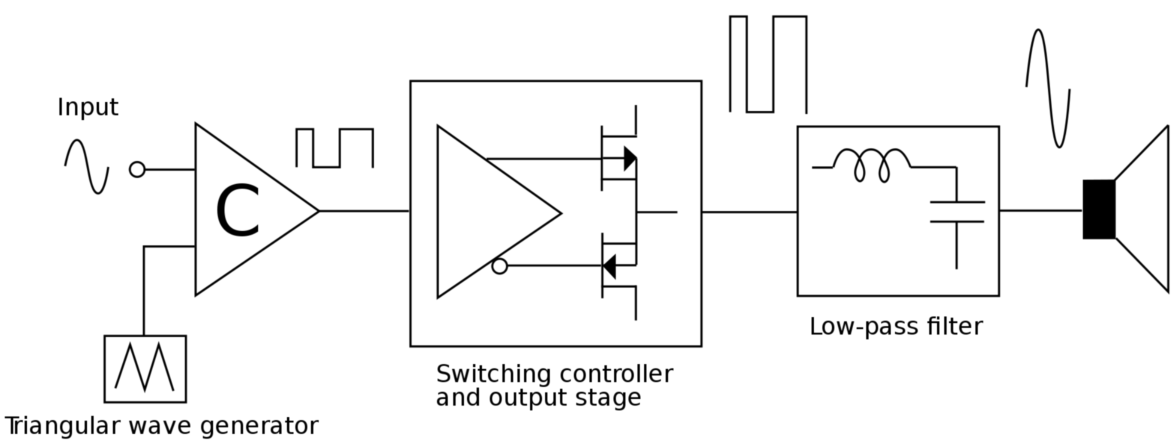


Figure 2.9.3.1: Single Stage Class D amplifier signal flow. Reproduced under a GNU Free Documentation License version 1.2

It is worth mentioning that although the final stage seems like a simple low-pass filter careful consideration must be taken when implementing it. If the filter is not sharp enough some of the carrier frequency in the 100 kHz's range will bleed through the speakers. Although humans might not be able to hear it and the speaker may not be able to reproduce it will try too. This can prove disastrous to the diaphragm and especially for tweeters and other components meant to be used for high frequencies.

Class H amplifiers

Class H amplifiers are just like class B amplifier designed to have a conduction angle of 180 degrees and to provide a two device push-pull output that also makes use of class D technology in the way of a Pulse Width Modulation scheme. The main addition to this particular amplifier class is that class H amplifiers use rail switching. That is, instead of the signal rising up to the level of a static power source rail, the rail switches its level with the signal. This increases the power efficiency of the amplifier tremendously. Efficiencies upwards of 90% are achievable (Self 34). In order to understand the concept of rail switching more easily figure 2.9.3.2 illustrates what an output signal with adaptive rails (upper ones for the positive bias, lower ones for the negative bias) looks like. It is worth mentioning that there are several readily available class H amplifier designs with single rail implementations. The TDA7396 audio amplifier IC used in this project was a Class H audio amplifier.

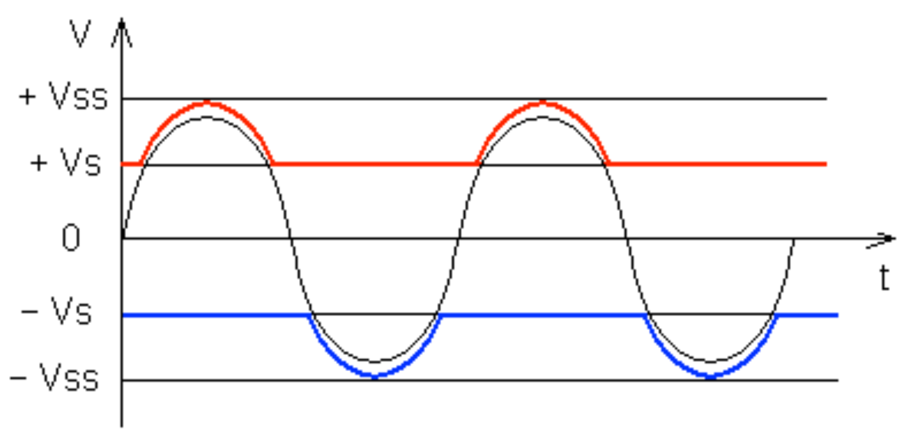


Figure 2.9.3.1: Class H amplifier output characteristic with rail switching Reproduced under a GNU Free Documentation License version 1.2

2.9.4 Noise

An audio amplifier sounds like a simple device to implement (pun intended). Just build an amplifier, either with Op-amps or transistors, that takes an input waveform and applies a gain higher than unity to all the electrical information situated between 20 Hz and 20kHz (the frequency band that humans can perceive with their ears). Nothing could be further from the truth. The physical limitations of the circuitry used for the amplifier design will always add a certain amount of noise and distortion to the amplified signal. In this context signal noise does not refer to any “hum” or “hiss” produced by a faulty connection or by improper grounding. Noise refers to unwanted voltage oscillations present at the output waveform of the amplifier. Notice that a distinction is made early on between noise and distortion, the reason for this is that noise depends on the physical construction of the individual components and it is physically impossible to completely eliminate it. Distortion on the other hand is inherent of the design and can be minimized to be even less than 1% of the total signal by a variety of means.

2.9.5 Thermal Noise

Noise can take the form of thermal noise, shot noise, flicker noise or burst noise (1) Shot noise occurs when the molecular energy of charge carriers varies as those charges are carried. Since the conduction of charge is unavoidable we will not account for shot noise in our design and we will instead focus in thermal noise. Thermal noise is temperature dependent as its name indicates. Also it is distributed evenly across the frequency spectrum. This means that the bigger the bandwidth of our op-amps, transistors and transformers, the bigger the amount of noise introduced by our resistors, capacitors and current sources. It is often calculated as an RMS voltage for each component. For example the noise generated by one resistor is:

(1) $$e = \sqrt{4kTR} \quad (1)$$

Where k is Boltzmann’s constant, T is ambient temperature in degrees Kelvin and R is the actual resistor value. The units of voltage noise in resistors are: $\frac{V}{\sqrt{Hz}}$.

This means that the bigger the operating bandwidth of the circuit the bigger the noise accumulated on it will be. In the same manner we can analyze the noise by a capacitor with the following relation:

(2) $$e = \sqrt{4kT / C} \quad (2)$$

Where C is the capacitance value. Notice that the units of capacitor noise are in Volts RMS. This is because capacitor noise is not frequency dependent (2); instead it is a constant noise floor within the circuit. It is obvious from equations (1) and (2) that the noise of the circuit is directly proportional to the bandwidth and the resistor value while it is inversely proportional to the Capacitance. Avoiding large resistors or unnecessarily small capacitors will minimize the amount of noise in the circuitry. Since noise is measured as a voltage and behaves linearly it can be added up by superposition of powers. The total noise going through the amplifier stages shall be calculated for the final design.

Noise from the power supply circuitry shall not be added up into these calculations. The reasoning behind this is that the power supply rejection ratio (PSRR) from the audio amplifier IC (TDA7396) is 60 dB. This will take out any excessive ripple from the DC voltage feeding the amplifier or the equalizer, preamplifier or mixer circuits. No actual line level or microphone level audio will make use of these lines anyway. Also the input common mode rejection ratio of the TDA7396 is of 70 dB . The design is up to par with modern technologies. For example the Roland Mobile Cube (citation) has a signal-to-noise ratio of 85 dB. This design project plans to match or surpass this spec.

2.10 Audio Interfaces

2.10.1 Audio Connectors and Cables

Audio connectors can carry balanced signals or unbalanced signals. Balanced connectors will have 3 different connection points or pins. They will be allocated for “Hot” or positive signal, “Cold” or negative signal and ground. The kind of cable that is used for these connectors is standard 2-pair shielded cable. The 2 pairs are used for the positive (or on-phase) signal and the other one for the negative (or inverted signal). The shield is used as the ground for both signals. 2-pair wire takes advantage of the common-mode-rejection-ratio that occurs as the wires are twisted inside the cable’s jacket. Since the wires are transmitting signals 180 degrees out of phase with each other and one is subtracted from the other at the end of the cable this cancels out most of the noise present in the cable. It essentially makes the twisted pair cable act as a differential amplifier. This makes this kind of cable configuration ideal for long cable runs. An unbalanced connector makes use of only 2 pins, One for signal and one for ground. Since this cable does not enjoy the common-mode-rejection-ratio that the

twisted pair does it is very prone to electromagnetic interference. To prevent it from picking up noise it should only be used for short runs.

Professional audio cables have 3 wires inside a plastic sheath. Two of the wires carry the same audio signal but 180 degrees out of phase with each other, this is done so when one signal is subtracted from the other at the final end of the cable the signals do not cancel each other out. The third wire acts as the ground connection and its pin is tied to the ground inside any black box that the cable is coming from or going to. To actually connect the audio sources such as guitars, MP3 players and microphones there are several connectors in the market. The design implemented in this project will make use of the XLR, the TRS and the mini-jack.

The XLR connector is shown in figure 2. Pins 1,2 and 3 are Ground, +Signal and –Signal respectively.

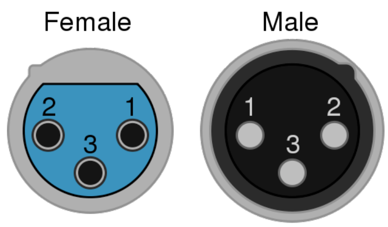
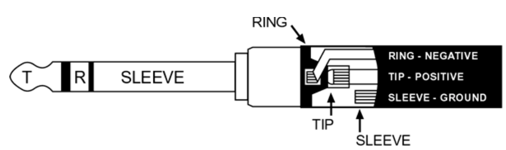


Figure 2.10.1 XLR connector pin-out. 1:1 scale. Reproduced with permission from Clark Wire & Cable

Instruments, MP3 players and computers make use of TRS connectors or mini-TRS connections. Their pin-outs are the same so only TRS connectors are discussed. The following shows the TRS or Tip, Ring, Sleeve configuration. By convention the tip holds the on-phase signal, the ring holds the negative signal and the ring acts as the signal ground.



2.10.2 TRS connector pin-out. 1:1 scale. Reproduced with permission from Clark Wire & Cable

Since the aforementioned TRS and XLR connectors are the standard kind of connectors for both instruments and microphones the design meets the requirements to interface with them. This is accomplished with the use of each connectors counterpart or in the case of microphones with a simple impedance

transformer that converts the low impedance output of the microphone with the high impedance output of an instrument. This impedance transformer will step up the low level signal from the microphone (2 mV RMS) up to 20 mV RMS by running it through a preamplifier with a voltage gain of 10. which will have to be mounted in the actual chassis of the amplifier. The *Unplugged* posseses a ¼ inch input and one 1/8 inch input that can accommodate one instrument connector and one auxiliary input/ MP3 capable connector. All of these connectors follow the same convention of pin 1 for the path to ground, pin 2 for the on-phase or "hot" signal and pin 3 for the out-of-phase or "cold" signal. Figure 2.12.3 shows which connectors shall be used to interface microphones and microphone cables to the actual circuitry of the amplifier.

1:1 Scale

1:1 Scale

Figure 2.10.3. Chassis-Mount XLR and ¼ inch TRS connectors. Reproduced with permission from Clark Wire & Cable

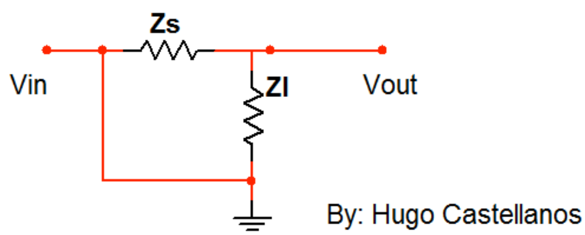
2.10.2 Impedance Matching and Bridging

Impedance matching seems to be one of the most misunderstood concepts of audio engineering. When talking about loads from a power electronics perspective the concept of matching a circuit's output impedance with a load's input impedance is necessary if maximum power transfer is desired. When talking about audio signals the exact opposite is desired. Devices input impedance must me much higher, which translates to at least 1 order of magnitude (Eargle, Foreman 89). A more proper title for this section would have been impedance (un) matching, because this is exactly what is needed when interconnecting audio components with each other.

Speakers and Amplifiers are designed as voltage driven devices. If both the output impedance of the input device (microphone, guitar, MP3 player) were matched with the input impedance of the actual audio amplifier circuitry this would "load-down" the voltage of the signal and generate a current equal to the amplitude of the input voltage divided by the input impedance of the device. This loading down minimizes the amount of voltage going into the input device (to the

amplifier) which is not convenient since all the voltages that are being amplified are rather low already in the order of 100’s of millivolts for line level signals and a couple millivolts for microphone level signals. The desired effect

Then it is obvious that the actual “matching” aspect of input and output impedances comes with impedance bridging. Let’s say that the output impedance of a voltage source is Zs and the input impedance of an audio device is Zl. When Zl >> Zs the effective impedance of Zs can be assumed to be 0 ohms so the totality of Vin would be dissipated across Zl. This would result in a maximum voltage transfer characteristic for the impedance bridge. This is exactly what is needed for ac voltage signals that are within the audio range. All the signals in the analog domain that are coming out of an instrument and into an amplifier and ultimately into a transducer have been subjected to careful analog signal processing. The possibility of an ac voltage signal losing any significant amount of voltage swing just because of simple interfacing is just not acceptable for pro audio applications. The aforementioned explanation is further illustrated in figure 2.10.2.



2.10.2 Impedance Bridging for Audio Components. Image by Hugo Castellanos

The unplugged audio amplifier has 2 inputs that have an input impedance of 10k. This assures that a variety of audio sources can be connected to it. Again, versatility is the name of the game for this project.

2.10.3 Pre-Amplification

Line Level Signals vs. Microphone level signals

There are two main kinds of levels for audio signals. These are Line level and microphone levels. Signals at line level are usually between components of an audio system such as between an MP3 player and a mixer console or between a TV and an audio equalizer. They are in the range of 100’s of millivolts and up. In the other hand electric signals generated by an audio transducer such as a

microphone or electric guitar pickups are in the range of 1.5-50 mV. This is to low for regular audio amplifiers. If a transducer with high output impedance is connected to an audio amplifier with also a high input impedance the connection will “load down” the circuit and would act as a voltage divider. This may reduce the signal level by up to 6 dB. To avoid this, the low level microphone or instrument signal is connected to a preamplifier circuit that would raise the signal level to the 100’s of mV range. As discussed in the impedance matching subsection the input of this pre-amplifier must be at least 10 times greater than the output impedance of the source feeding the signal. Line level signals in the other hand do not require any pre-amplification. They can be fed directly into the audio frequency amplifier. This implementation is done in the Unplugged by running the Microphone/Instrument input through a preamplifier with a voltage gain of 10 and running the mp3 player input directly into a 2 band equalizer that is then fed into a unity gain mixer that combines both the mp3 player and the microphone/instrument signals.

A more convenient way of differentiating line level signals from microphone level signals is the dBu. It is a decibel unit where the reference voltage for the ratio is 0.775 Vrms and the output load is 600 Ω. The 0dBu point will happen when the load dissipates 1mWatt of power. Line level signals are referred to be at +4 dBu and Microphone and Instrument level signals at -10 dBu.

2.11 Audio Equalizing

2.11.1 Background

The main goal of equalization in the product is to compensate the signal deficiencies of each input among the audio frequency spectrum. This allows the user to shape the sound within reasonable parameters by increasing the audio gain of different frequency bands. This audio equalizing chain is implemented as an active filter.

Active and passive implementations offer distinct advantages and disadvantages. The main advantages of active filters are that they can easily be implemented with operational amplifiers and few resistors and capacitors; they can deliver high quality factors with high accuracy. The disadvantages of using active filters are that most op-amps that can be used for them such as the NE5534 require a dual sided Voltage source of at least 12 Volts (Datasheet 1) that would draw anywhere from 1.7 mA to 2.8 mA depending on the ambient temperature. This amount of current is given per each individual op-amp. In the case of the

LME49740 the current load is about 20 mA. This sounds really high when compared to 2.8 mA, but this 2.8 mA is for a single IC. The LME49740 offers us 4 IC's which are implemented as one preamplifier, one Baxandall tone control,one unity gain buffer and a 3 input summing amplifier at the heart of the project.

Another possibility is to use a passive filter implementation. In a nutshell taking this route would make use of more components. The potential disadvantages of this translate in the potential need for a bigger PCB than one with active filters. In the plus side the fact that there are no Op-amps reduces the power consumption of the circuitry as well as the possibility of introducing noise in the signal via the op-amp's power supplies. Both avenues were considered and an active filter implementation was deemed better for the product.

2.11.2 Audio Filters

The preliminary schematic of this project projected a 3-band "cut-only" equalizer. The term 3-band could lead to misleading. Audio equalizers divide the audio spectrum in several bands. When audio equalizers talk about boosting or cutting bands they are actually referring to the center frequencies of those bands. The idea of having "cut-only" filters comes from the reasoning that all the signal amplification should be made by the audio amplifier and the audio amplifier only. If the equalizer is able to boost the circuitry it might drive the voltage swing beyond the rails of the audio frequency amplifier and cause distortion. Also, using a passive audio filter would have reduced the number of total IC's and would increase the possible amount of current drawn by other peripheral devices such as the LCD screen, the audio effects processor and the power management unit.

The simplest implementation of a passive filter is of a passive T-Networks for each desired frequency. Once a frequency band is defined, say between 25 Hz and 625 Hz a passive T network is implemented by using fixed-value resistors and capacitors in the following configuration.

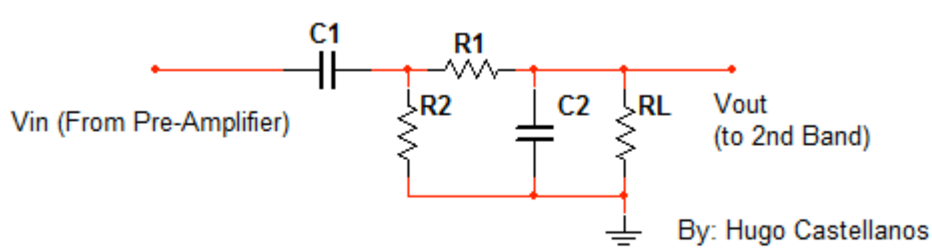


Figure 2.11.1 Passive Band-Pass Filter

Implementing a single filter with these parameters would be extremely simple. Each frequency of interest could be tuned with variable capacitors housed in the front-end of the device. The main disadvantage of this design is that in order to implement 3 frequencies three different filters should be cascaded. In order to avoid each circuit loading down each successive circuit a non-inverting unity gain buffer should be put in between each circuit. Since the final design is a 2-input device (microphone/instrument and MP3 player) it was foreseen that at least three 3-band equalizers would have been implemented so it would have cost 6 op-amps total. This amount of operational amplifiers would incur in a significant expense in dedicated boost converters for each one of them. The alternative to this is to use active circuitry to handle the equalization duties.

Since the specifications of the project only called for a single equalizer circuit for the MP3 player, one of the amplifiers of the LME49740 was deemed as the equalizer for the whole device.

Shelf Filters

A compromise between using purely active or purely passive audio filtering resides in the baxandall tone control topology. In this topology one active component like a transistor or an operational amplifier divides the frequency spectrum in two around a single center frequency. The implementation for the design is a 2-band active filter called a baxandall tone control. The baxandall tone control offers a shelf kind of equalization where a low-pass filter and a high pass filter affect all the frequencies of the spectrum except a middle one where they meet. For illustration a typical Baxandall frequency response is included in figure 5.

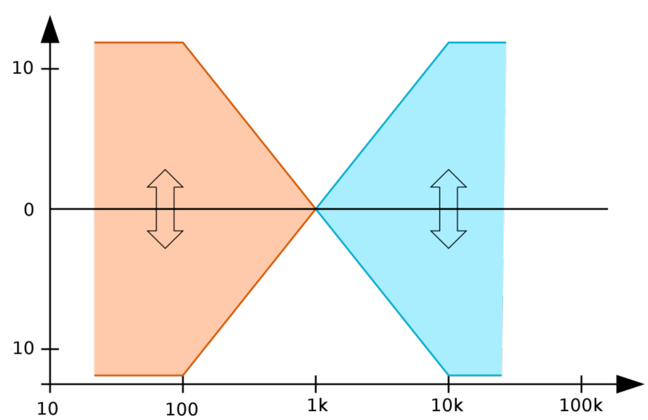


Figure 2.11.5. Baxandall Tone Control Gain(dB) vs. Frequency(Hz) graph. Reproduced under GNU free documentation license version 1.2.

3. Design

3.1 Component selection

3.1.1 Power Supply

For this research solar panels are going to be used to charge a battery which is going to be used to power the whole system. To choose the right solar panel, many things were taken in consideration. First, all the current requirements for all the components and parts of the system were calculated. For the microcontroller a current of 500uA was required to power it, whereas for the DSP controller 100mA were needed. Also, the LCD display uses 300uA, the virtual ground of the audio system used 400 µA, preamplifier 4.6mA, each of the two regulators consume 15mA, the amplifier consumes 90mA, and the equalizer only consumes 20mA. Adding all this currents and assuming some leakage occurs gives a total current of around 300mA.

To find the current and time it takes to charge a battery completely the following equations were used.

$$Current\ =\ Watts/Volts$$
$$Time\ to\ charge\ =\ Current \times Time/\ Current$$

The type of material of solar cells is a very important characteristic of solar panels. By taking in consideration that a 12V rechargeable battery is going to be used to power the system without the need of plugging it, a 12V, 20 watts solar panel is chosen as the main source to charge the battery.

The solar panel chosen has 36 multicrystalline cells in series which produce 20 watts and 12 volts. The solar panel and its IV characteristics are shown in figures 3.1.1.1 and 3.1.1.2.

Solar Panel Features Found in [15]:

Open Circuit Voltage (Voc): 21.6V
Short Circuit Current (Isc): 1.3A
Maximum Power Voltage (Vmp): 17.2V
Maximum Power Current (Imp): 1.17A
Dimensions: 420x420x25 mm

Figure 3.1.1.1 Monocrystalline Solar Panel.

Modeling of Solar Panel

To verify the behavior of the solar panel chosen with different parameters the Shockley diode equation was simulated in Matlab. It is a simple model that has a diode, resistance in series and photo current source as it is shown in figure 3.1.1.2.This model has solar irradiance, temperature and voltage dependence.

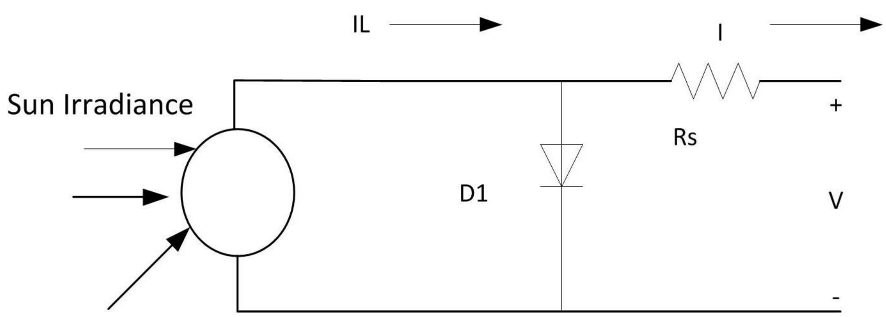


Figure 3.1.1.2. Circuit used to simulate the solar cell.

In this circuit the diode is used to find the IV characteristics of the photovoltaic cell.

The following equations found in [16] were used to find these characteristics:

$$I = IL - Io(e\hat{}q((V + IRS)/nKT) - 1)$$
$$IL = IL(T1) \times (1 + Ko(T - T1))$$
$$IL(T1) = G \times ISC(T1)/G(nom)$$
$$Ko = (ISC(T2) - ISC(T1))/\ T2 - T1$$
$$Io = Io(T1) \times ((T/T1)\hat{}3/n) \times (e\hat{} - q \times Vg/n \times K) \times (1/T - 1/T1)$$

$$Io(T1) = ISC(T1)/(e^{\wedge} q \times VOC(T1)/n \times K \times T1 - 1)$$
$$Rs = -dV/dIVOC - 1/XV$$
$$XV = Io(T1) \times q/(n \times k \times T1) \times (e^{\wedge}q \times VOC(T1))/n \times k \times T1$$

Several variables in these equations are constants determined by the manufacturer of the solar panel used. These values are shown in table 3.1.1.1. At full sunlight the value of irradiance is going to be around 1000W/m^2.

The Schockley diode equation is used to find the relationship of the current and output voltage in the absence of light. The saturation current was found using open circuit voltage and short circuit voltage of the solar panel given in the solar panel datasheet.

Used variables from datasheet	At T = 25 C	Units
VOC	21.6	V
ISC	1.17	A
Max Power	20	W
Voltage at Pmax	17.2	V
Current at Pmx	1.17	A

Table 3.1.1.1: Main characteristics of the solar panel.

Because of the dependence of temperature and irradiance in the equations the different effects of these parameters can be shown in two different graphs. Figure 3.1.1.3 shows the effects of changing the irradiance value, whereas figure 3.1.1.4 shows how different temperatures changes the voltage output of the solar panel.

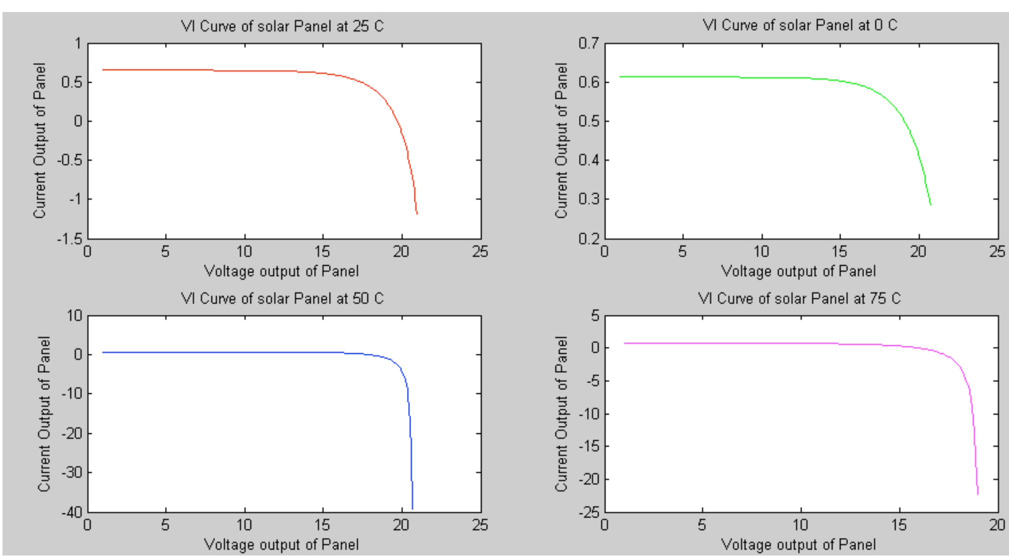


Figure 3.1.1.3 Voltage output of simulated solar panel with half of the sun irradiance.

This figure shows the direct effect that the amount of irradiance has on the output response of the solar panel. It is seen that the amount of output current is greatly decreased; therefore, the power of the solar panel is going to decrease as well. On the other side as it can be seen in figure 3.1.1.4, the voltage of the solar panel decreases when temperature rises.

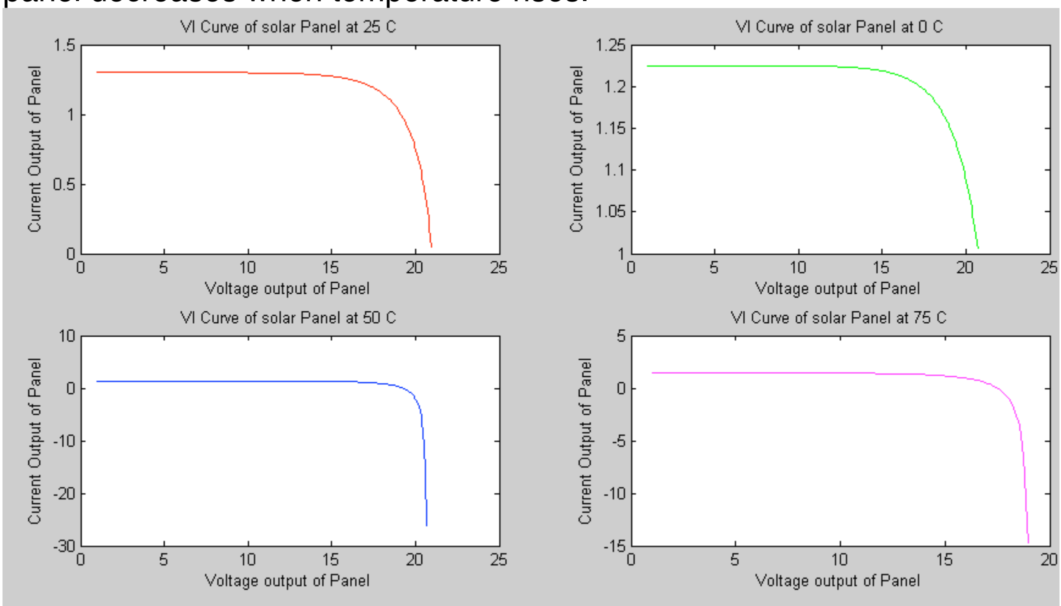


Figure 3.1.1.4 Voltage output of simulated solar panel with different temperatures.

Also, the solar panel was simulated using Simulink with only one dependent variable, temperature as it is shown in Figure 3.1.1.5

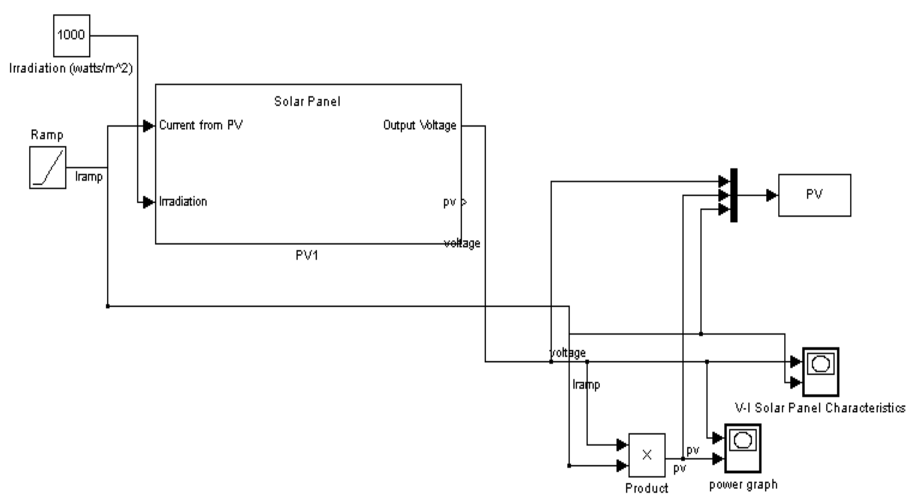


Figure 3.1.1.5 Simulink Simulated Solar Panel with Irradiance 1000 Watts/m^2. HQRP solar panel parameters were used.

With these simulations not only the voltage output in figure 3.1.1.6 was found, but also the output power of the solar panel as shown in figure 3.1.1.7

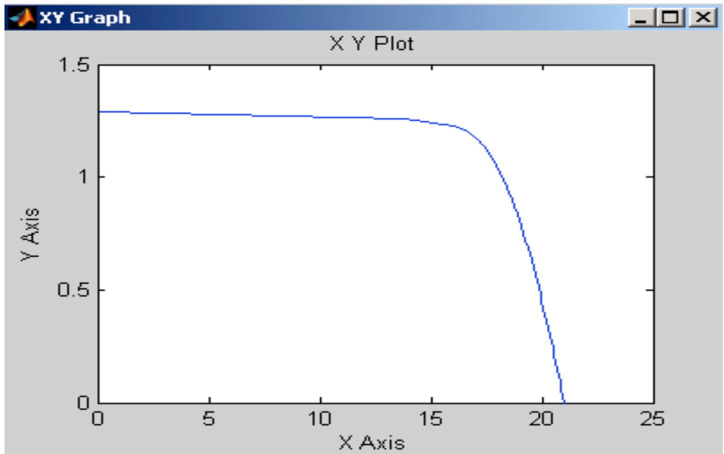


Figure 3.1.1.6 Voltage output of simulated solar panel Voltage vs. Current

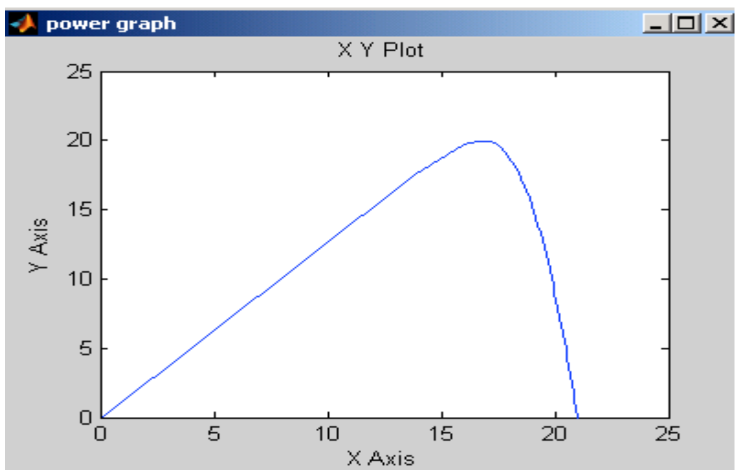


Figure 3.1.1.7 Power output of simulated solar panel Voltage vs. Power. This figure shows clearly that the maximum power of the solar voltage is 20 Watts and that it occurs at 17.2 Volts.

3.1.2 Battery

The battery is one of the most important components of the system. It is going to be used to power up the microcontroller, the LCD, and is going to be the provider of energy to the whole audio system.

A couple of batteries were compared and it was decided to use a Sealed Lead Acid battery because it can be charged by using constant voltage and is less expensive than other batteries. The battery model is shown in figure 3.1.2.1. This type of batteries performs well at high temperature and can be charged for long times.

Figure 3.1.2.2 Power-Sonic 12V/5AH Sealed Lead Acid (AGM) Battery

One of the goals of this project is to make the system work with a battery for at least 3 hours so lead acid battery of 5.0 Ah was taken in consideration. Using the battery datasheet figure 3.1.2.1 shown below, for a 1A current gives the system at least 3 hours of power use.

Battery Specifications Taken From [14]:

Valve regulated allows safe operation in any position
Measures L: 3.54 in W: 2.76 in H: 3.98 in
High Shelf Life (% of nominal capacity at 68°F (20°C))
1 Month 97%
3 Months 91%
6 Months 83%
Weights 3.5lbs
Energy Density (20-hr. rate) 1.54 W-h/in3 (94.16 W-h/l)
Specifi c Energy (20-hr. rate) 17.14 W-h/lb (37.79 W-h/kg)
Internal Resistance (approx.) 40 milliohms
Max Discharge Current (7 Min.) 15.0 amperes
Max Short-Duration Discharge Current (10 Sec.) 50.0 amperes
Operating Temperature Range
Charge .-4°F (-20°C) to 122°F (50°C)
Discharge -40°F (-40°C) to 140°F (60°C)

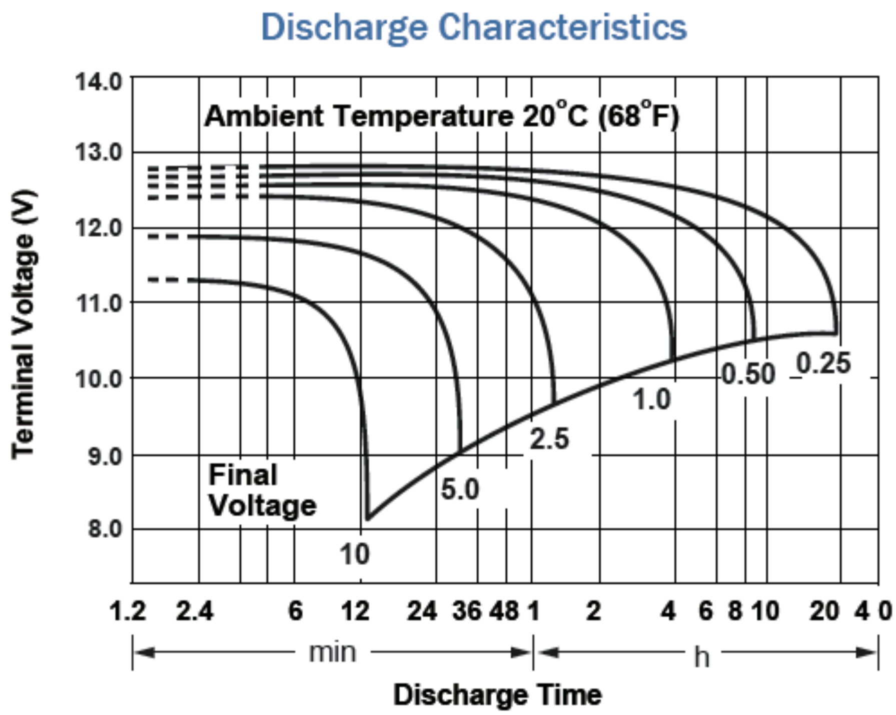


Figure 3.1.2.2 Lead Acid Battery Discharge Time. Reproduced with permission of PowerSonic.

3.1.3 Microcontroller

The most important part of the monitoring subsystem is the microcontroller. The main purpose of the microcontroller is to gather information from the solar panel, battery, and effects unit to later communicate with the LCD in order to display solar panel and battery voltage, battery remaining percentage and current audio effect.

This microcontroller is turned on by a voltage regulator, which is powered by a battery, allowing the microcontroller to work even when there is no sunlight received by the solar panels. It was important to take in consideration the amount of inputs and output pins of the microcontroller and verify that they were sufficient for the design to be implemented completely, as both the battery and solar panel outputs are read by the microcontroller and displayed in the LCD.

The Arduino Uno board uses an ATmega328 chip. The following list shows some of the characteristics of this microcontroller:

1 Kbytes EEPROM
23 digital input/output pins
6 PWM16 Analog inputs
16 MHz crystal oscillator

32 KB flash memory
USB connection
Input/output pin 40 mA of current.

One huge advantage of using this microcontroller is that it can be programmed using C language. Aside from this, it also has 23 analog and digital input/output ports, which are more than enough for the design. This microcontroller also has a pin, which can be used for voltage reference, and this was important to the project as this pin was connected to 5V as a reference so that the reads from the battery and the solar panel are more accurate. On figure 3.1.3.1, the Arduino Uno board is shown.

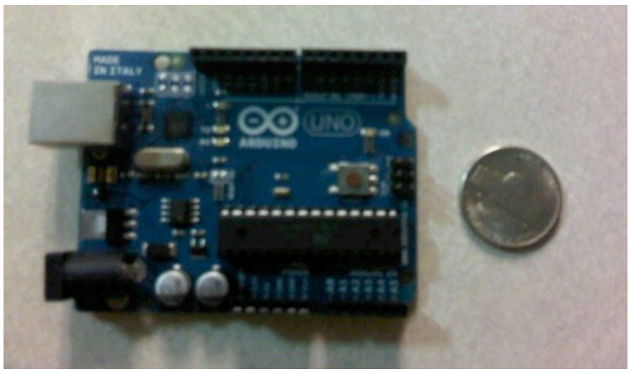


Figure 3.1.3.1: Arduino Uno Board.

3.1.4 Display Screen

The group considered two options for an LCD display that were explored on section 2.10. Both of these LCD screens presented advantages and disadvantages that directly impacted the project. Both of these displays also met the project's required functionality, so it was not a simple decision when one of them had to be chosen. The biggest factor that influenced the choice of LCD display was price and availability.

The chosen LCD Display for this project was the Hitachi HD44780 LCD panel. Given that four values are displayed on the LCD, the 16 x 4 size was chosen. The specific model chosen was Longtech's LCM1602T character LCD Module. Many factors influenced this choice. One of the first factors considered was price. Staying within budget was one the main concerns when choosing all the components for this project. The HD44780 LCD panel fully meets this requirement as it runs for about $7 on a 16 x 4 size. This is a great price for a display screen. Another reason that steered the decision towards the LCM1602T LCD panel was availability. This LCD screen can be found in many Internet

stores, including eBay. What this meant for the group was that the screen was ordered and received within a week. This was a great advantage for this project, as time was very limited and the soonest the parts were received, the better.

One of the team's main concerns when choosing an LCD display was that it could be easily interfaced with the chose microcontroller. The LCM1602T allows for either a 4-bit interface or an 8-bit interface, which provided flexibility when choosing the microcontroller. Initially during the research phase, it was considered that two microcontroller would be used, one for the power management system and one solely to control the LCD Display. This idea was quickly abandoned after examining the cost and complexity that having two microcontroller on the system would bring. It was finally decided that only one microcontroller would be used for both the power management system and the LCD screen. When this was decided, it became imperative that the LCD selected was compatible with this microcontroller. From the performed research, it was clear that the HD44780 LCD display was compatible with the power management microcontroller, which in changed influenced the team's decision for this display.

The HD44780 LCD panel is not only compatible with the Arduino Uno, which was selected for this project; it is widely used with this brand of microcontrollers. Arduino website has a section dedicated to this part. It helps by giving methods, code samples, and interfacing instructions. This was a great tool for the team since no one had programmed an LCD through a microcontroller before. This section and the many other blogs and websites that instruct students on how to use this part greatly influenced our decision to choose the LCM1602T. Another one of the aspects that were important when making a decision on the LCD display was a vast use of the part. Through research conducted for this part, it was discovered that this LCD display has been widely used in many types of projects. Most of the projects using the part were student projects similar to this project. This was also a decision point considered by the team. From all the types of projects observed during research, it could be seen that this LCD could be easily programmed an interfaced to a microcontroller.

The last factor that influenced the team's decision was low power consumption. It has been emphasized during this project that low power consumption is one of the main goals for this project. This amplifier runs on battery power most of the time, so it is imperative that all the components chosen for this project have very low power consumption. The team's goal is to maximize battery life, and this cannot be achieved if close attention is not paid to the component's power

consumption. The LCM1602T has an operating voltage of either 3.3V or 5.5V. It also has an operating current of 1.5 mA. If the backlight is used, it has a voltage of 4.2V. These voltages and currents fit into the power budget do that the battery runs for the desired time. The block Diagram of the LCM1602T is pictured below.

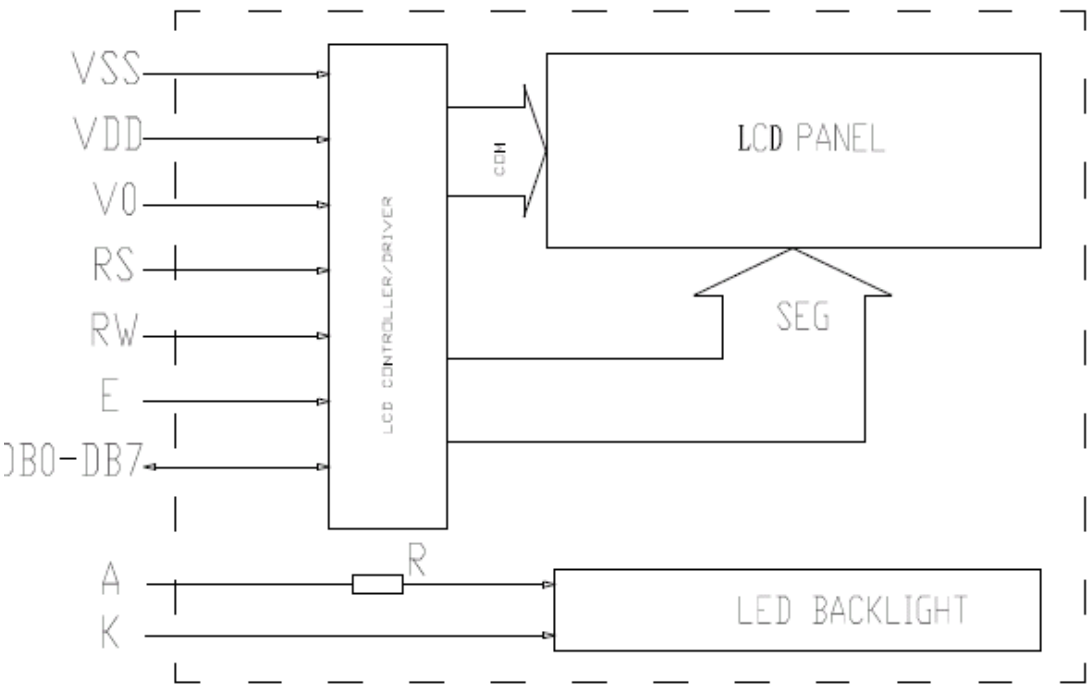


Figure 3.1.4.1: LCM1602T Block Diagram. Reproduced with permission from Longtech Optics.

From the block diagram, it can be seen that the LCM1602T has 16 pins. The pins are explained briefly below.

1. VSS is the ground pin
2. VCC can be +3.3 or +5V. For the effects of this project, 3.3V will be chosen.
3. VO is contrast adjustment pin.
4. RS is the register select. RS=1 is Data. RS=0 is command.
5. RW is the read/write pin. When R/W is equal to 0, the LCD is in write mode. When R/W is equal to 1, it is in read mode.
6. E pin is the clock. This is a falling edge triggered device.
7. Pins 7-14 are the LCD bits that will be used to interface with the microcontroller.
8. A pin is the backlight anode.
9. C is the backlight cathode.

In conclusion, this part was chosen because it seemed to have the best fit with the project's specifications. It met and surpassed the budget requirement. I also met the power requirement that specified the lowest power consumption possible. The part is also readily available, and there are many programming resources available to students working with the piece. Finally, the part has been used extensively and successfully in many types of student projects. All these reasons drove the decision of choosing the LCM1602T as the display screen for the project over the previously researched option, the Nokia 3310.

3.1.5 DSP Component

The decision on a DSP component for this project was not an easy one. There were many factors that had to be taken into consideration when making a choice for the DSP component. Like the rest of the components in the project, the DSP component had to meet some basic requirements to be chosen, which included price, low power consumption and availability. For the decision on this component, there were some major factors that also had to be taken into consideration when making the decision. These factors were programming complexity and interfacing. After considering all of the factors mentioned above, there was one DSP component which was a clear choice for this project.

The DSP component chosen to use in this project is the Belton BTSE-16FX audio module. The BTSE-16FX was chosen for this project for several reasons. The first reason why this audio module was chosen was the fact that it included reverb, which was one of the effects that the project was specked to implement. It not only includes reverb, it also includes 2 types of delay, 4 types of chorus, and two types of flanger. It was also chosen because in order to implement the effects, no other external hardware was required, as the other components mentioned on the research section required either an EEPROM or a microcontroller in order to operate. Since there are 16 effects that are included, a grey code rotary encoder was used in order to select the desired effect. Another consideration when making this choice was interfacing with the microcontroller. It was important that the effects unit could interface with the microcontroller in order to display the audio current audio effect. With the BTSE-16FX, interfacing was simple by using 4 pins that connected to 4 of the ATmega's digital pins.

3.2 Design Implementation

3.2.1 Battery Charger Circuit

For the battery charger the monolithic step down battery LT3652 IC was chosen because it provides maximum peak power tracking and is specifically designed for solar energy purposes. It also provides constant-current/constant-voltage charge and current can be programmable up to 2A. This current can be programmed by using a sense resistor with a maximum voltage drop of 100mV [13].

$$Rsense = .1/Ichg(MAX) \qquad (1)$$

The cost of this IC is inexpensive compared to other MPPT modules that require more complex circuits with discrete components such as MOSFETS and the use of a microcontroller with pulse width modulation programming. It only requires a minimum set of components such as regular resistors, a sensing resistor, diodes and an inductor.

To make the solar panel operate at its peak power the LT3652 was programmed using a voltage divider in its input. This voltage divider maintains the solar panel at the peak power voltage. To program the MPPT function equation 2 was used. Vin is the voltage of the solar panel including the voltage drop of the input diode and Vin_Reg = 2.74V.

$$R1 = (Vin/2.74V - 1) \times 100K\Omega \qquad (2)$$

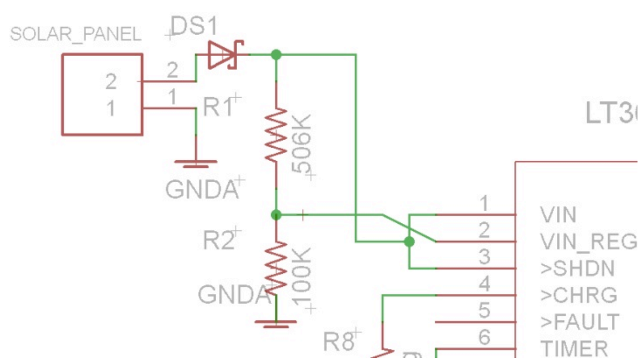


Figure 3.2.1.1. The input divider connected in the Vinreg pin of the chip the LT3652 is able to make the solar panel work at its MPP.

To program the battery float voltage to 14.4 V the IC uses another resistor divider connection. A thermistor was used to compensate for battery voltage changes that occurs with temperature change. The equations used for this configuration are (3), (4).

$$RFB3||(RFB1 + THERMISTOR||RFB2) = 3.3V/14.4V - 3.3V \quad (3)$$

$$RFB4 = 250K\Omega - [RFB3||(RFB1 + THERMISTOR||RFB2)] \quad (4)$$

By setting RFB3 to 100KΩ and using a 47KΩ thermistor the value of RFB2 was found to be 75KΩ. The 3.3V used on equation 2 comes from the voltage on the VFB pin and the 250KΩ on equation (4) is used to compensate for input bias current error [13].

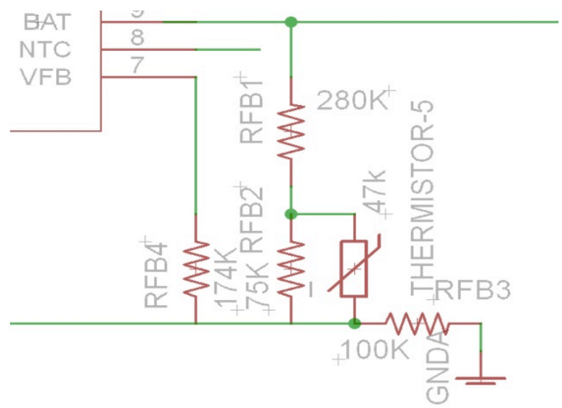


Figure 3.2.1.2. The configuration used to program the float voltage of the battery.

The maximum current was programmed to 1A, once the battery reaches C/10 it terminates the charging process which will prolong battery life and avoid overcharge. At this state the LT3652 will draw less than 1 μA from the battery. The charger will enter the fast charging mode once the battery goes below 13.2V and trickle charge at .15A if it drops below 10V [13].

The full circuit shown in figure 3.2.1.4 is a Power Point Tracker which is used to maintain the solar panel at its maximum power voltage.

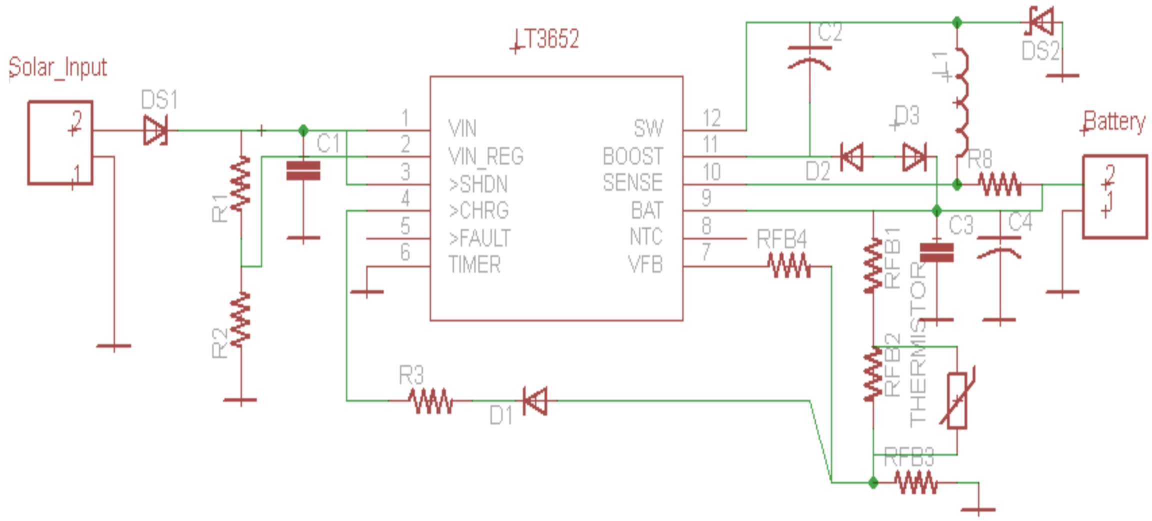


Figure 3.2.1.3: PPT Battery Circuit using the LT3652.

A Maximum Power Point Tracker can increase the efficiency of the solar power circuit. To verify how the MPPT works the IV curve of the solar panel is plugged in. As an example figure 3.2.1.4 is going to be used to describe how this works. This graph shows the big drop in current that occurs when the current reaches highest voltage of the solar panel. The current is constant, but decreases at that specific point.

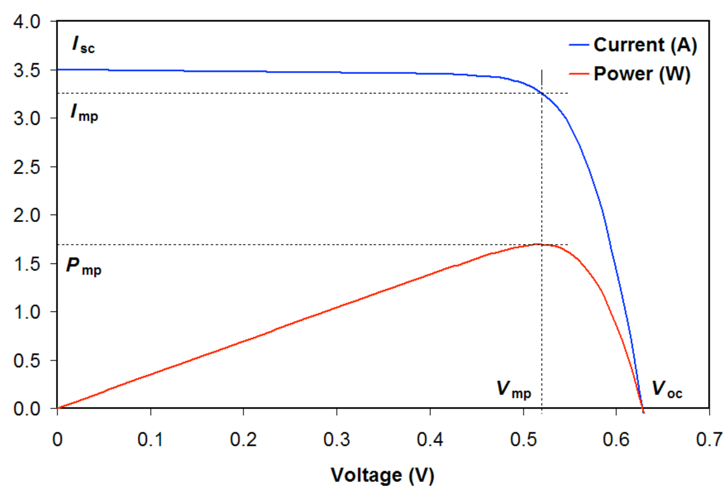


Figure 3.2.1.4: Example of a Solar Panel IV Curve with MPP. Reproduced under GNU free documentation license version 1.2.

The most important part of this graph is the solar power output of it is shown in the bottom line. This power is measured in Watts, which is the current times the voltage created by the solar panel. For this example shown the solar panel needs to provide around .52V to produce the highest amount of Watts, which is the Maximum Power Point. Using a solar panel to charge a battery is going to lower

this voltage because the battery will want the solar panel to work at its voltage [13]. For a 12 V battery the solar panel voltage will drop to 12 Volts and this will reduce the amount power greatly compared to its Maximum Power Point value. This is why the use of MPP is so important to keep the solar panel working at its maximum to get more output power.

By using a monolithic step down battery charger IC the maximum peak power voltage of 17.2 V outputted by the solar panel can be converted to the 12 V of the battery. This IC is needed for the circuit to reach the MPP. It changes the higher voltage of the solar panel and the lower current to the voltage and current required to charge the battery. It lowers the voltage and reduces the current. The same watts that are input are outputted with the use of the LT3652 at a different voltage and current.

Maximum Power Point of solar panels is not fixed because it changes with the amount of irradiation and temperature as it is shown in the figures 3.2.1.5 and 3.2.1.6 below. This figures show how the maximum power is greatly reduced with irradiance. With half the irradiance the power is reduced by half. The first figure shows the power output when the irradiance is at its maximum 1000 W/m^2 and the second one is at half of that irradiance. They are also different for other solar panels.

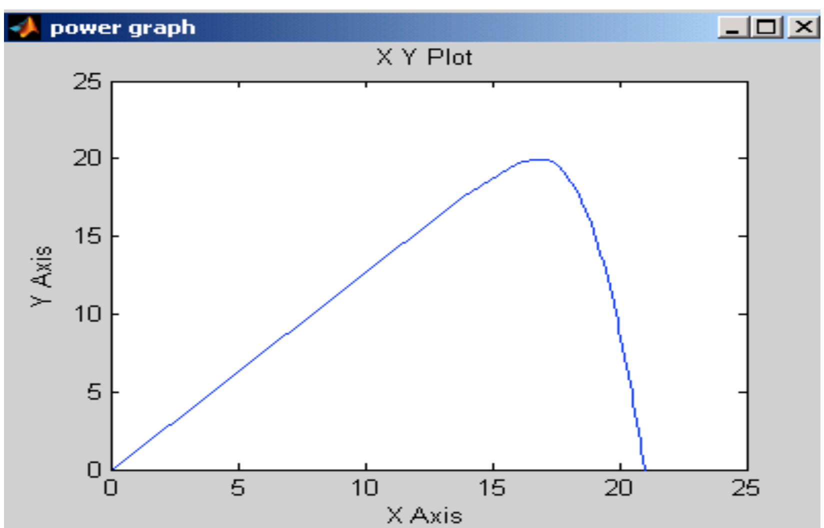


Figure 3.2.1.5: Power output Voltage vs. Power of simulated solar panel for 1000 W/ m^2 of irradiance.

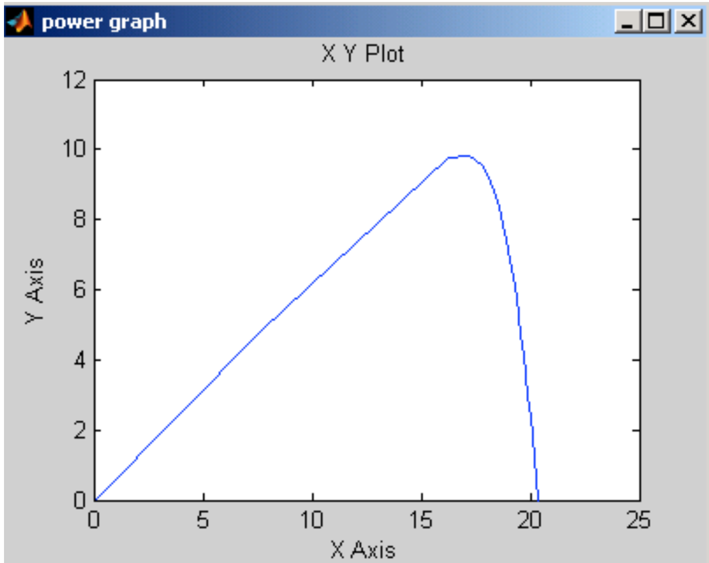


Figure 3.2.1.6: Power output of simulated solar panel at half of irradiance Voltage vs. Power.

In order for the DC/DC converter to work correctly ratio of the input voltage to output voltage needs to be changed according to the MPP of the solar panel chosen.

The main purpose of this PPT is to change the high and low current of the DC/DC converter to low voltage and high current for the power of the battery charger.

The MPP voltage is directly related to the temperature; hence, when the temperature of the solar panel increases the MPP voltage decreases as it was shown in previous a graph which means that the solar panel MPP voltage is going to be very similar to the battery voltage with the increase of temperature.

3.2.2 Power Distribution

For the power distribution a direct connection topology was chosen. This implementation was chosen over path selection topology because it was easier to implement, less expensive, and did not need switching networks. The solar panel was directly connected to a battery charger. The load and the charger were separated by two switches located at both ends of the battery. This was so the system could be used only in charging mode, load mode or both modes together. Also, two DC-DC converters were used to step down the Voltage from the main 12V SLA battery. A 9V regulator the LM2940T was used to power up the audio amplifier, mixer, equalizer and preamp, while the effects unit, the LCD, the Atmel

328 microcontroller required a 5V source which was implemented by using a LM2931 5V regulator. These regulators were chosen because they provide low noise which is a requirement for the audio design and low dropout voltages of 500mV and 450mV respectively which make them ideal for the entire design.

3.2.3 Monitoring Subsystem Block Diagram/Schematic

Figure 3.2.3.1 in this section corresponds to the block diagram of the microcontroller and the information that it received from its corresponding inputs. As it is observed in this figure, the microcontroller receives three main inputs, battery, solar panel, and effects unit. In order for the microcontroller to read solar panel and battery voltage, two analog pins were used, each one reading one of the inputs respectively. When it came to the effects unit, four digital pins of the Atmega328 had to be used in order to identify the current effects since there were sixteen different combinations. Finally, the microcontroller runs an algorithm in order to determine battery remaining percentage based on the battery voltage that it reads. Once all of this information is read by the microcontroller, the four values are displayed to the LCD display in order for the user to monitor the whole system.

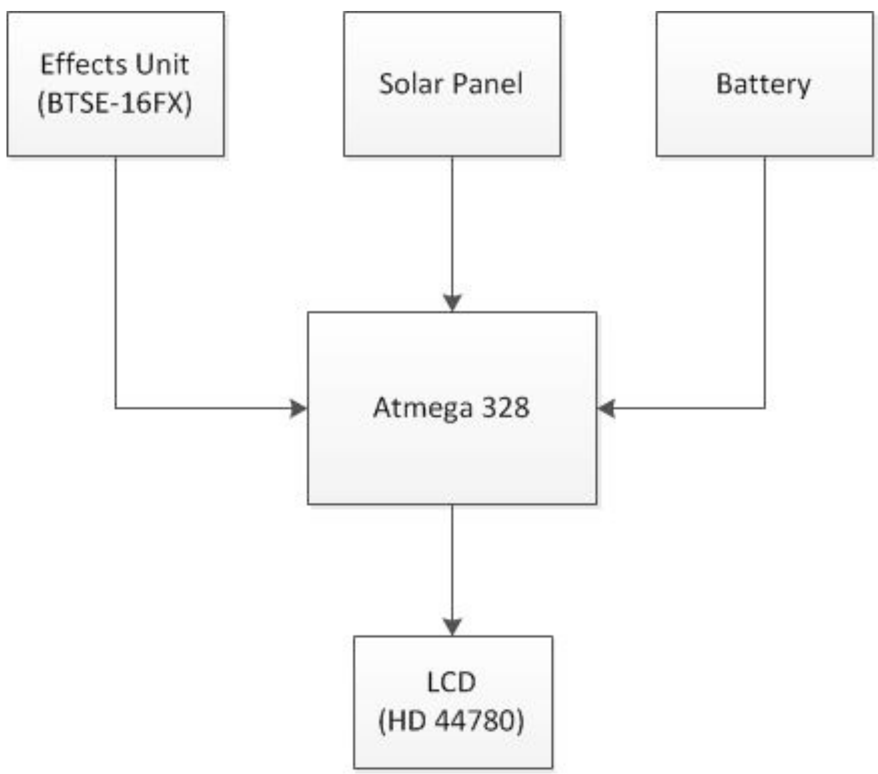


Figure 3.2.3.1: Monitoring Subsystem Block Diagram

As it was mentioned before, the microcontroller and the LCD are the main components of the monitoring subsystem since they are used to monitor the system internally in order to inform the user of the state of the system. Below on figure 3.2.3.2 is the schematic of the monitoring subsystem. As it can be observed from the schematic, six of the Atmega328 digital pins, as well as VCC and GND are used in order to interface the microcontroller with the LCD. Even though they are not pictured here, the solar panel input uses analog pin 0 and analog 5 for the microcontroller to read battery and solar voltage respectively. In order to interface the effects unit with the microcontroller, four digital pins were used in order for the microcontroller to read one of sixteen different combinations and then display the current effect to the LCD. In order to select the audio effects, a sixteen position grey code rotary encoder was also needed. From the schematic, it can be observed that the encoder is connected to the same pins on the BTSE-16FX that are read by the microcontroller. All of the information is constantly monitored by the microcontroller and then updated and displayed to the LCD for the user to see.

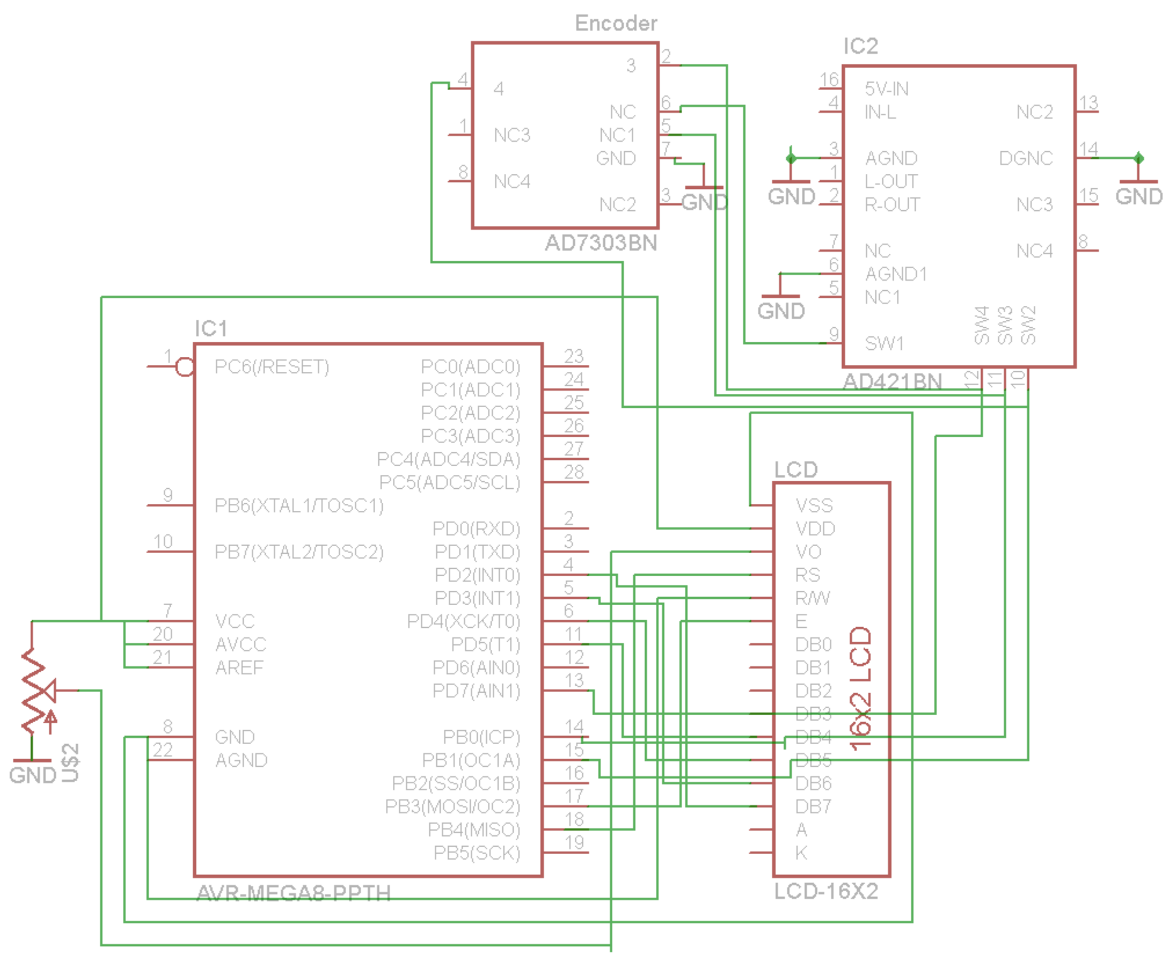


Figure 3.2.3.2: Monitoring Subsystem Schematic

3.2.4 Audio Subsystem

Special Design Considerations

A particular characteristic of the actual equalizer design is that it does not operate with dual rail operational amplifiers. This is because of the battery powered nature of the Unplugged. This makes the realization of dual supply operational amplifiers particularly difficult since all devices that use dual rail voltages do so by center-taping a stepdown transformer from the 120 V AC mains and using that point as a reference voltage. Designing the system to be powered from the AC mains would have added bulky transformers, power capacitors and a myriad of buck and boost converters for each block of the device. This goes completely against the main goal of the device which is to be an easy to transport amplifier for outdoor use. The solution to this problem is the use of a virtual ground with an operational amplifier. A virtual ground essentially takes the Vcc source and splits it in half and feeds it through an operational amplifier in a buffer configuration. The fact that the operational amplifiers attempts to maintain its input voltage equal to its output voltage generates a very precise voltage reference that can be used as a ground point. In the case of both the baxandall and pass-band filters, the non-inverting input of each operational amplifier is tied to the output of this virtual ground. A feature that is always present in virtual ground implementations is the presence of rather large capacitors in the non-inverting input of the buffer as well as a large resistor pair. This assures that the quiescent current fed trough the amplifier is very small. We must remember that the voltage source for all this circuitry is the 12 Volt 5.0 ampere-hour lead acid battery. Because of this a TLE2426 virtual ground from Texas Instruments also known as "The Rail Splitter" will be used. This virtual ground is extremely appropriate for this application because it only requires a 170 uA typical operating current. For a clearer understanding of the virtual ground to implement the topology for the circuit is given in figure 3.2.4.1.

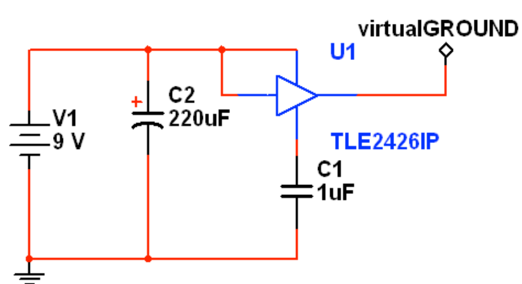


Figure 3.2.4.1 Virtual Ground Circuit. Image by Hugo Castellanos

Audio subsystem design overview

In order to provide the reader with a deeper understanding of the audio subsystem and the different circuits that conform it figure 3.2.5.2 is provided. It showcases the 2 possible different inputs of the device (MP3 or Instrument/Microphone). In the case of the MP3 player it is ran through the Equalizer. The Instrument/Microphone signal is ran through the Preamplifier instead and brought from a nominal level of 2-20 mV to 20-200mV. This signal in the range of 100's of mV is ran through the DSP Effects unit. All 3 inputs are summed by the mixer and finally are ran through the audio amplifier and delivered into the speaker load.

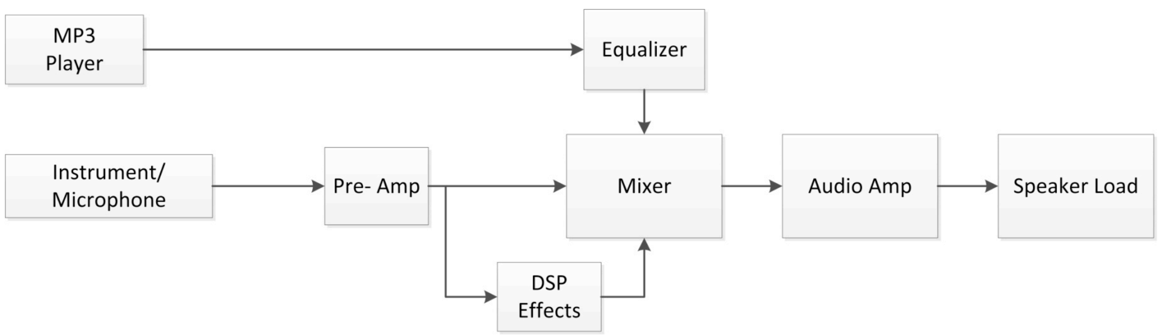


Figure 3.2.4.2: Audio Subsystem Overview

Preamplifier Considerations

In order to minimize the number of IC's the equalizer scheme that was used in the amplifier is a 2-band baxandall tone circuit. Due to cost limitations this circuit was limited to affect the mp3 player input of the Unplugged. Another device to consider is the Preamplifier for the microphone/instrument input of the device. Since this is an active device its frequency response is of upmost importance to the final implementation of the Unplugged sound system.

There are 3 possible frequencies/behaviors that can be taken into consideration. The reasoning behind picking these 3 frequencies as the center frequencies of the 3-band equalizer was done by experimenting with actual music. The iTunes® music player was used to audition different music genres such as rock, Latin and jazz. This particular music player has a built-in virtual 10-band equalizer with frequencies at 32 Hz, 64 Hz, 125 Hz, 250 Hz, 500 Hz, 1kHz, 2 kHz, 4kHz, 8kHz, and 16kHz. The dynamic range of each frequency is +/- 6dB. In layman's terms this means that a frequency in the audio spectrum can be made 2 times as loud

or quiet with respect to the other ones. After listening to several different audio samples in personal computer speakers and headphones it was determined that the majority of bass content is centered around 125 Hz. Also, the majority of "midrange" sound information was determined to be around the 1kHz frequency. In order to determine the 3^{rd} and high frequency for the equalizer a linear scale was used. Since 1kHz / 125 Hz = 8, the third frequency was placed at F_3 =8*1kHz = 8kHz.

A first attempt to implement a band-pass filter for each frequency was done with the help of Texas Instruments' document "Filter Design in 30 Seconds" (Carter 6). The given circuit is a dual supply op-amp with the following dual rail topology:

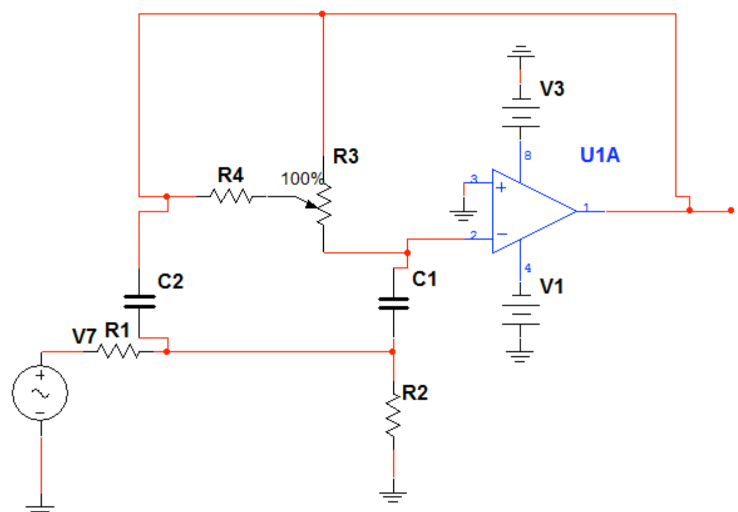


Figure 3.2.4.3 Narrow (Single Frequency) Band Pass Filter

The design procedure is outlined as follows:

Procedure 1: Narrow Band-Pass Filter

a) Pick a capacitor value $C$and make $C = C_1 = C_2$.
b) Calculate $R1, R4. R1 = R4 = \frac{1}{2*\pi*C1*Fc}$
c) Calculate $R3 = R1 * 19$
d) Calculate $R2, R2 = \frac{R1}{19}$

The values were picked to implement a 125 Hz band-pass filter. Final values were approximated to standard 5% resistor values:

a) $C1 = C2 = 0.01uF$

b) $R1 = R4 = \frac{1}{2*\pi*C1*Fc} = 127324\Omega \approx 127k\Omega$

c) $R3 = 127\ k\Omega * 19 = 2.43 * 10^\wedge 6\Omega \approx 2.4M\Omega$

d) $R2 = \frac{127k\Omega}{19} = 6684\Omega = 6.8k\Omega$

The resulting schematic can be seen in figure 3.2.4.4.This design shall be referred from now on as the low-end filter.

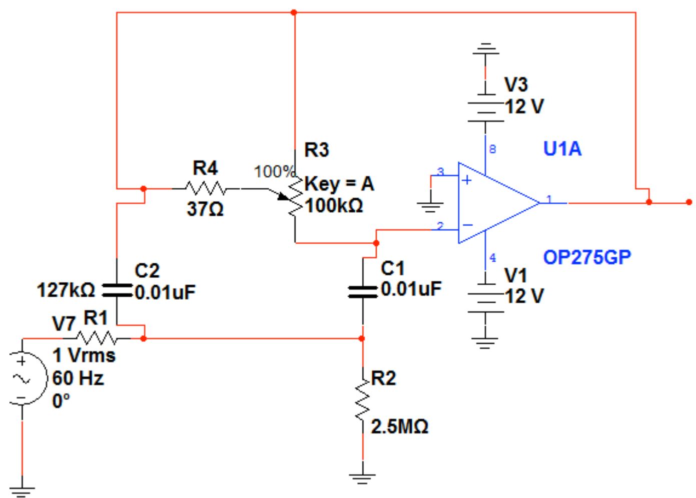


Figure 3.2.4.4 Narrow (Single Frequency) Band-Pass with f_c=125 Hz

The frequency response was tested within the human audible range (20Hz-20kHz). And he bode plot in figure 3 was obtained. The plot is between 20 Hz and 20 kHz and the gain range of +/- 50 dB. The cursor is at 126.832 Hz with a recorded gain of minus 5.569 dB:

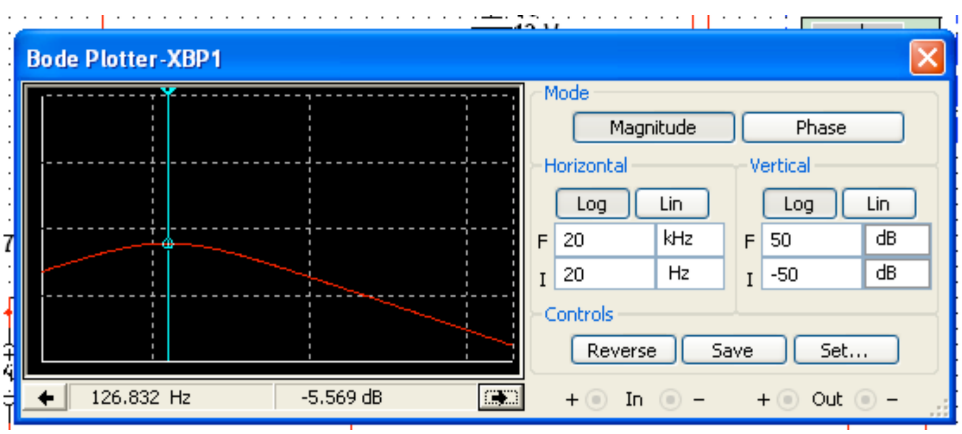


Figure 3.2.4.5 Frequency Response of Low-end Filter

The small variation in center frequency from 125 Hz to 126.832 Hz is deemed responsible to the approximation of resistor values. This approximation is deemed acceptable for the scope of this design project.

Another consideration for the implementation of the Low-end filter is the phase response of it. Since all of the 2 possible input signals (Microphone/ Instrument and MP3 player are going through identical inverting chains (the equalizer in the case of the MP3 and the Preamplifier in the case of the instrument/microphone) no phase cancellation is observed in the implementation of the device. the chance of phase distortion between the signals is minimal. Also, since the system is monophonic (only one output speaker, no stereo image) there is no way in which signals of different amplitudes but opposite wavelengths would cause subtractive interference cancellation. Nevertheless, in order to better understand how the filter behaves a closer look at the phase response is given. Figure 4 shows the phase response between 20Hz and 20kHz with a magnitude of 180 degrees.

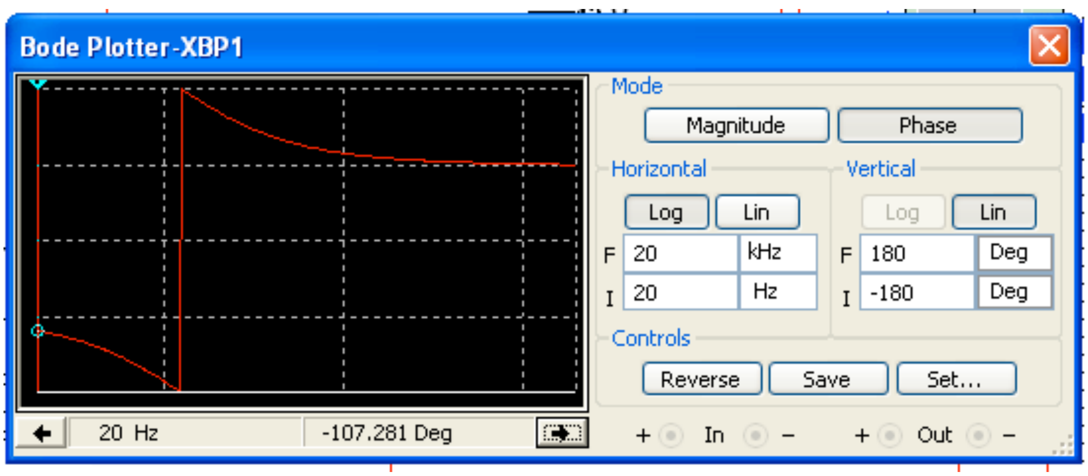


Figure 3.2.4.6 Phase Response of Low-end Filter

As expected figure 4 shows a drastic phase-shift around the center frequency of 125 Hz. There are not foreseeable problems since the 3-band equalizer block is feed-forward only. At no point in the system will the original audio input signal and the filtered input signal be reproduced together. If this happened there would be all sorts of comb-filtering, possible amplitude cancellation and even phase distortion.

In order to use this filter in the equalization chain a way to increase or decrease the gain of the center frequency must be implemented. This is accomplished with the use of a potentiometer in the feedback loop. In order to place it where it can

control the gain of the amplifier we derive the transfer function of this filter. For the sake of brevity we do not include the derivation of it. The inclusion of this gain control is actually what will allow the user to set the gain of the filter and to make up for acoustic deficiencies of the speaker, the room or even the instruments or transducers being fed through it. Also, the circuit is modified so its non-inverting input and its negative bias input are now connected to the virtual ground.

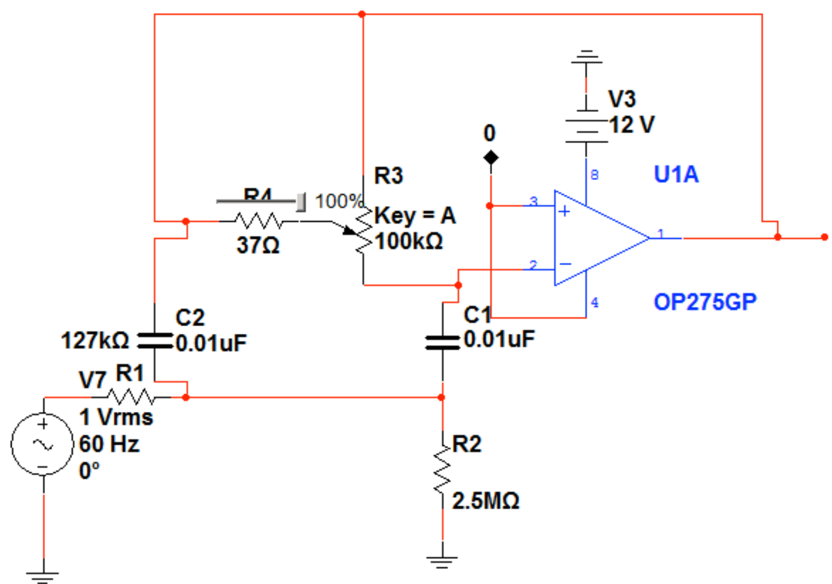


Figure 3.2.4.7 Low-end Filter with Gain Control

Final Implementation

In order to design the audio amplifier we must take into consideration the nature of the signals connected to it. A typical dynamic microphone is capable of converting a human voice 1 meter away from it at 80 dB SPL levels into an AC voltage signal of 2mV. In the other hand a commercially available electric guitar such as a Fender Stratocaster© has pickups capable of transforming an acoustic wave into an AC voltage with a peak to peak amplitude signal with a range of 70 mV to 200 mV. Since the *Unplugged* is a device primordially designed to be used to play music priority was given to signals in the 20mV to 200 mV ranges. The audio preamplifier consists of one of the operational amplifiers within the LME49740 connected in the following configuration:

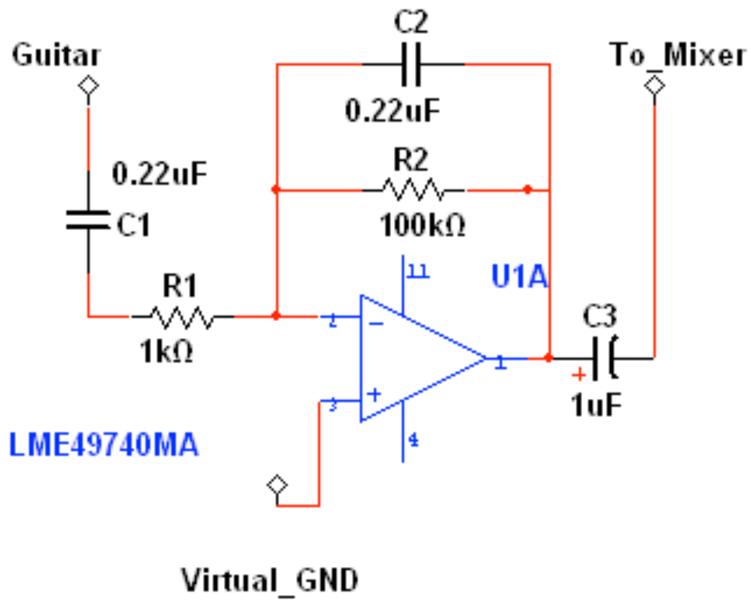


Fig. 3.2.4.8. Audio Preamplifier with an inverting gain of 100. Notice the added film capacitors for stability. The 1uF output capacitor acts as a coupling capacitor between this circuit and the mixer.

As it can be seen in Fig.3.2.4.8, the preamplifier inverts the instrument input. The C1 0.22 uF capacitor will filter the DC component of the input. The manufacturer recommends the capacitor in the feedback path for added stability to the circuit. Both the input and feedback capacitors are film types due to their reduced noise figure in the audio band.

Baxandall Tone Control

The Baxandall tone control topology is available in passive and active configurations. Previous experimentation by the team with the passive Baxandall circuits did not deliver expected results so an active Baxandall topology was favored. The one obvious difference between the passive and the active implementations of the baxandall filters are that the active configuration can actually deliver gain. Careful consideration must be put in picking the values of the baxandall circuitry so that the user does not clip the audio signal because of too much envelope shaping with the equalizers. The total gain of the Baxandall circuit will be limited to +/- 6 dB of amplification or cut. The active baxandall topology for the actual design can be appreciated in figure 3.2.4.9.

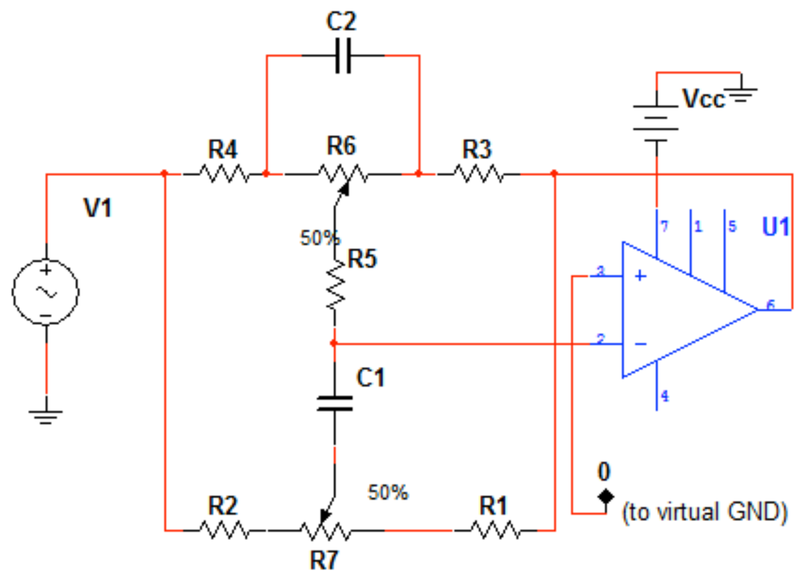


Figure 3.2.4.9 Active Baxandall Filter Topology . Original design by Ramon Vargas Patron from INICTEL, Lima, Peru.

The design equations for the active Baxandall filter topology are as follows:

$R_4 = R_3 = R_5, R_1 = R_2$

The first corner frequency of the bass control potentiometer (R6) is given by:

$f_{low} = \dfrac{1}{2\pi C_2 R_6}$ with a shelve corner at $f_{low2} = \dfrac{1}{2\pi C_2 R_3}$

and the first corner frequency of the treble control potentiometer (R7) is given by:

$f_{treble} = \dfrac{1}{2\pi C_1 R_7}$ with a shelve corner at $f_{treble2} = \dfrac{1}{2\pi C_1 R_5}$

The center frequency of the circuit is defined by the relationship:

$f_{Center} = \sqrt{f_{Treble} * f_{Low}}$

The decision to place the center frequency of the baxandall EQ for this particular project was taken by considering the actual frequency response of the human ear. It is widely accepted and known that humans can only hear within the range of 20 Hz and 20kHz right after they are born. Continuous exposure to noise degenerates the human ear. According to a recent study by the University of Washington, some adults might even have their top hearing range reduced to 8 kHz. This is why for the design portion of this project the exact middle of the

audio spectrum band (around 10kHz) will not be used as the center frequency of the Baxandall. Instead the baxandall will be centered somewhat at 500 Hz. This will assure that the midrange of the equalizer is not affected by the tone controls and only the highs and the lows of the frequency spectrum are affected. Remember that the idea of the tone controls is to compensate for acoustic inadequacies of the room and not to completely shape the sound of the audio input. Calculating the components for a high shelf eq with a 3 dB frequency of 2000 kHz and a low shelf eq of 125 Hz would give us the values of table 3.2.4.1

Component	Value
R_1	33kΩ
R_2	33kΩ
R_3	22kΩ
R_4	22kΩ
R_5	22kΩ
$R_6 = R_7$	100kΩ (potentiometers)
C_1	800 pF
C_2	12.7 nF

Table 3.2.4.1: Component values for the Baxandall Equalizers

These values were simulated in Multisim and the following magnitude responses were obtained for different potentiometer positions. The first was the response of the signal magnitude when all of the potentiometers are in the middle position or 50% of turn (no gain, no cut). As it can be seen the response is completely flat between 20 Hz and 20 kHz. This is the expected result and it shall be implemented in the final design.

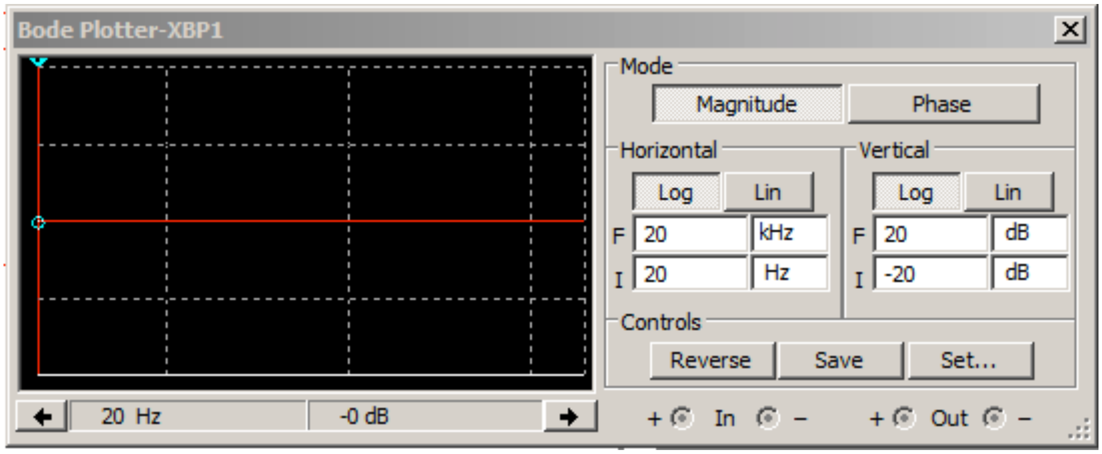


Figure 3.2.4.10 Baxandall Magnitude Response with potentiometers at center position (unity gain)

the piece of resistance of the baxandall tone control comes when actual boosting or cutting of the signal takes place. With the potentiometers placed at the 0 position (completely turned counterclockwise) the following output curve is obtained from the circuit. This agrees with the designed specification which was a steady slope gain cut from 584 Hz towards 20 Hz and a steady slope gain cut from 584 Hz on to 30 kHz. The maximum gain cut is 6 dB which is more than enough to compensate any room inadequacies or unwanted instrument frequencies. Figure 3.2.4.11 provides a graphical explanation of the aforementioned.

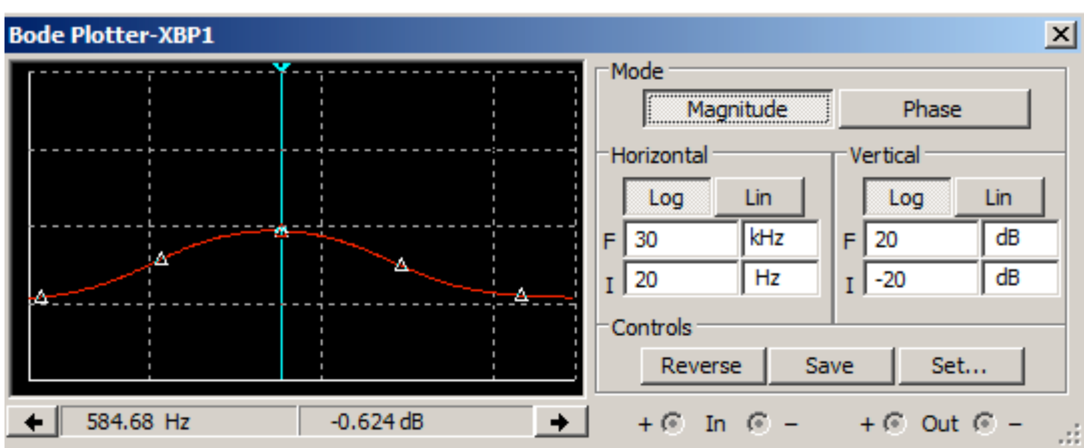


Figure 3.2.4.11 Baxandall Magnitude Response with potentiometers at maximum attenuation

And last but not least when the potentiometers are both completely engaged (rotated completely clockwise) the characteristic of figure 9 is obtained. As it can

be seen the maximum voltage gain for the treble frequency band is 5dB higher than the curve for the low-end bands. This asymmetry in practicality is irrelevant since the users will adjust the tone of the audio signal in accordance with their sound connect and no their electrical characteristics. The important thing is that the center frequency of the baxandall circuits remains the same as well as the constant slope of both the high end and low end bands.

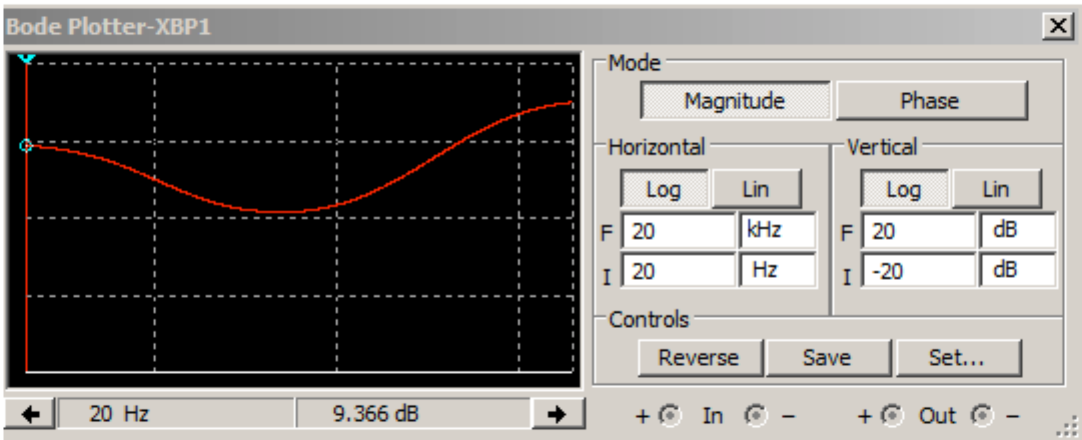


Figure 3.2.4.12 Baxandall Magnitude Response with potentiometers at maximum gain

Audio Mixer

The audio mixer consists of the 4th and last remaining op-amp of the LME49740 IC configured in an inverting configuration. Since the BTSE-16FX Unit has a built in unity gain buffer that will invert the instrument input channel, an inverting amplifier as a mixer assures that both the audio input are in-phase and it avoids any comb-filtering by the final acoustic signal delivered by the speaker. Also, it will not add any gain to the audio signal of the iPod or of the Instrument level input. Instead it will just combine the inputs from the iPod, the instrument/microphone input and the input from the effects unit into a single signal. The LME49740 operational amplifier is unity stable so this configuration does not pose any particular technical difficulty. The figure 3.2.5.11 depicts the final configuration of the audio mixer with the inputs from the preamplifier, the effects box and the auxiliary input of the mp3 player.

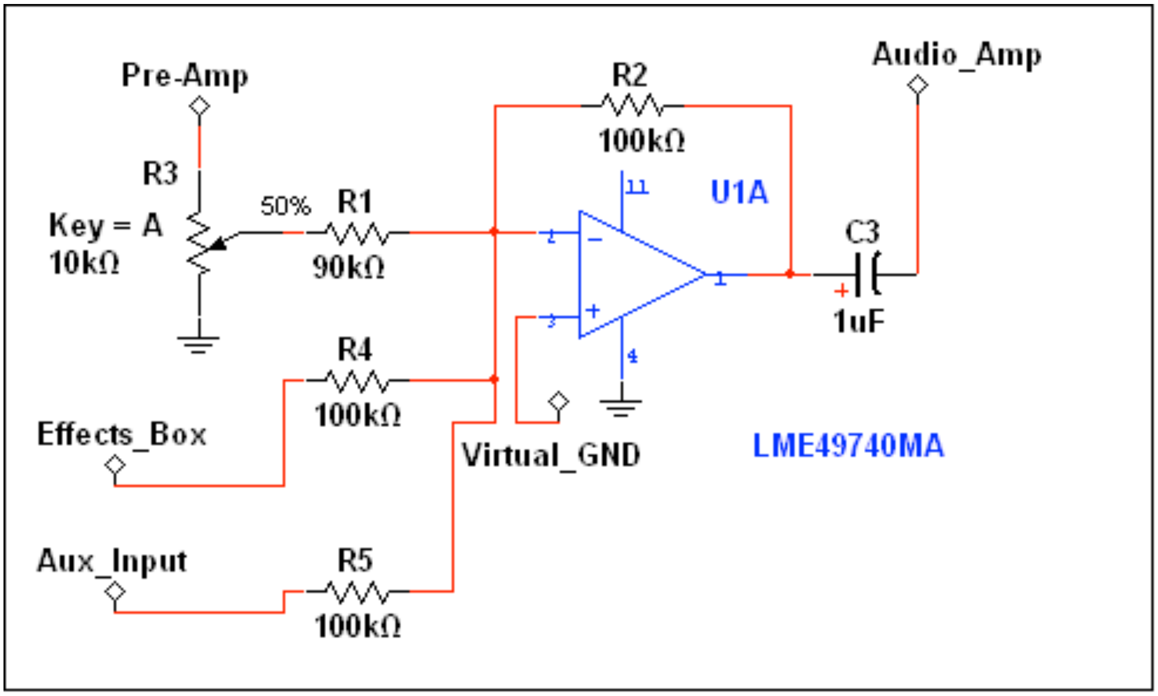


Figure 3.2.4.13: Mixer Configuration

3.2.5 Audio Amplifier

The audio amplifier IC used in this project was the TDA7396 from ST Microelectronics. It consists of a self- contained 45-Watt audio amplifier. It is capable of operating with a wide supply voltage ranging from 8V to 18 V while providing a constant gain of 26 dB. It is capable of amplifying fully differential inputs or single-ended ones like the ones used for this project. The fact that it has a low quiescent current of 100 mA and a high voltage gain of 26 dB make it ideal for the *Unplugged.*

This amplifier will be an ideal candidate for a battery powered system like ours because of its single-rail design as well as its inclusion of internal pre-amplifiers that can be used for low line level signals. This pre-amplifiers can translate in lower biasing voltages for other operational amplifiers such as the ones present in the Bandaxall equalization circuit, the equalizer or the mixer.

A very attractive feature of the TDA7396 is that it operates in class H mode until it detects a fault in the supply voltage. If the voltage drops suddenly the amplifier will start to operate in class B mode Also this amplifier will be more than capable of providing sufficient power amplification to the signal of both microphones, the instrument input and the iPod/Auxiliary source input. A detailed schematic of each stage of the amplifier can be seen in figure 3.2.5.1.

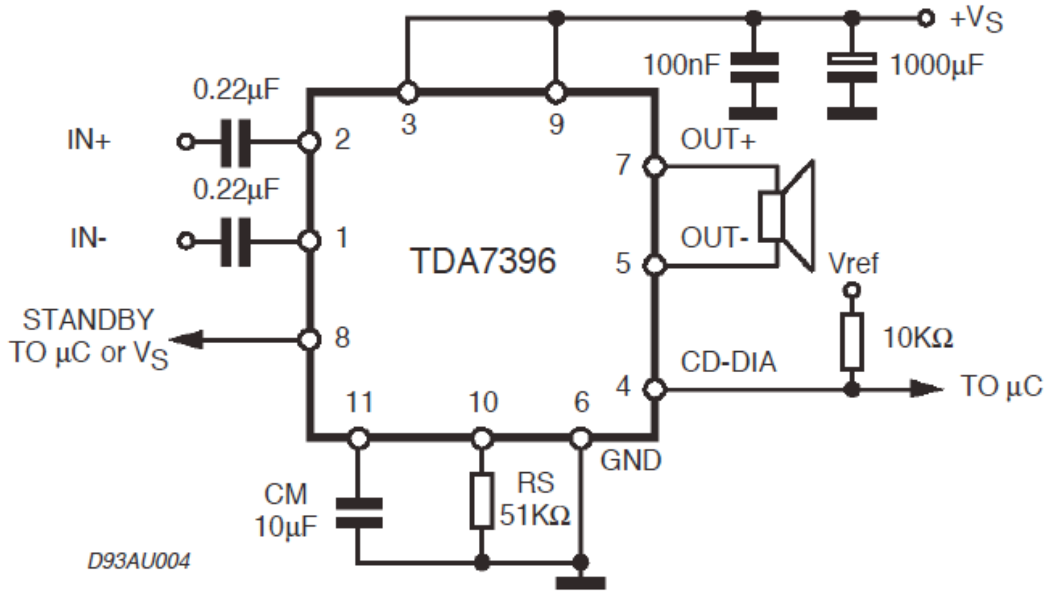


Figure 3.2.5.1: Internal configuration of the TDA7396 Printed with permission of ST Microelectronics.

3.3 Microcontroller Coding

The microcontroller coding was performed using the Arduino IDE which is provided for free from the Arduino website. The programming was also performed in C since it was a known language for all the team members. The final code was a mix of internal functions to read from the analog and digital pins and compute values and built in functions in order to display values to the LCD. To make the programming of the microcontroller simpler, Arduino provides a library of standard functions that were used in this project. The functions that were used in the programming of the Atmega 328 in this project were:

pinMode()- This method is used to configure a specific pin to function as an input or output. This function will be used to read data from the solar panel and battery and set the data received to be outputted to the LCD.

analogRead()- This function was used to read the values from the solar panel and battery voltage which were connected to connected to analog pins.

digitalRead()- This function was used to read the values fed onto the effects unit by the grey code encoder in order to display the appropriate effect on the LCD.

delay()- The delay function was used to pause the program for an specific amount of time so that the output displayed to the user was readable.

liquidCrystal()- This function that was used to create a variable of liquid crystal type. It takes in as a parameter either 4 or 8 data lines. It also takes in the

number of the pin that is connected to the RS pin on the LCD, the pin on the Arduino that is connected to the RW on the LCD, and the Arduino pin that is connected to the enable pin on the LCD. This function was indispensable as it was the one that allowed the LCD to display characters on the screen.

begin()- this function was used to specify the dimensions of the LCD screen.

display()- was used to turn on the LCD after it had been turned off. The function did not take in any parameters but it did require a variable of liquid crystal display type.

noDisplay()- was used to turn the LCD off but saving whatever is currently displayed on the screen. The function did not take in any parameters but it did require a variable of liquid crystal display type.

print()- Print was the most commonly used function since it was used to print text to the LCD.

There are many more functions that are provided by Arduino's libraries, but these functions were the only ones that were used for this project. As it was mentioned earlier in this section, there are also functions that were not built in and that were coded by the team in order to read in the values that came into the microcontroller. These functions were:

read_data()- This function did not take in any parameters. It was used to read solar panel and battery voltage, run some logic to convert the values to meaningful information and then save this information into two global variables.

print_data()- This function was written so that the solar panel and battery voltage could be displayed to the LCD. It took care of formatting the data correctly and also setting the LCD cursor in the right position so that this data to be displayed into the first two lines of the display. This function did not have to take in any parameters since the read_data() function stores the information into global variables.

print_effect()-Print effect was written to accomplish several tasks. First of all it read from four different digital pins that were connected to the effects unit and stored that information into variables. Once the information was read, this function ran an algorithm to determine which effect was currently being implemented. Lastly, this function displayed the current effect to the LCD.

stateofCharge()- this function was written because it was important for the user to know what the battery state of charge is. This function runs one of two algorithms to determine the remaining battery in form of a percentage based on the battery voltage that was previously read by read_data(). It needs to run two algorithms because it makes a difference whether the battery was currently

charging or not. When the battery is charging it is on an excited state and it will show a higher voltage than when it is not charging.

3.4 Overall Design

The *Unplugged* audio system is a very linear implementation with minimum signal feedback. Both microphone level and instrument level audio signals are amplified up to the line signal level before being fed into any circuitry.

This is the first analog signal treatment of the microphone and instrument level signals. The only audio input that has a separate component is the iPod/Auxiliary input goes through a passive purely resistive “stereo-to-mono” converter in order to add both the on-phase component and out-of-phase component of that audio signal. Then and only then is the iPod signal fed through one of the amplifiers of the LME49740 IC.

The second stage of analog signal treatment comes in the form of the dedicated tone bass and treble controls present at the MP3 channel strip. They are implemented with ¼ of the LME49740 operational amplifier with a single rail supply voltage of 9 Volts and 2 100kΩ linear potentiometers. The maximum gain/boost of the signal with this tone control circuit is +/- 6 dB. It is important to mention that the audio amplifier present at the last stage is designed as a single-rail pre-amplifier, but the LME49740 is not. In order to implement a battery powered single rail version of a Baxandall tone circuit a virtual ground was implemented with a rail-splitter op-amp, the TLE2426. All operational amplifiers that use a split rail power supply will be implemented instead with this virtual ground.

At this point the audio signal from each channel strip can be routed in several ways. One or more of the inputs can be fed into the Belton Engineering 16-FX audio signal processor where it can be treated with one of the 16 dedicated effects implemented by it. After being treated it leaves the analog output of the Belton 16-FX Chip and is fed back into the main signal flow path. If a particular signal from a particular channel strip is treated or not with the Belton 16-FX in the Effects Loop parallel to the main signal flow, it will regardless be added to the sum of the remaining analog audio paths. The signals will be added in an inverting amplifier configuration implemented by ¼ of the LME49740. The single ended signals are combined into one single signal and fed into the TDA7396 audio amplifier will then drive this mix down of all 3 channels and increase it by up to 26 dB delivering a power of 45 Watt RMS into a 4 Ω load. The beauty of the

TDA7396 is that this constant gain of 26 dB is maintained as long as the rail of the IC is kept above 8 Volts. This makes it an excellent choice for this design. The 4 Ω speaker load is a full range driver Yamaha 15 inch speaker cabinet connected to the TDA7396 output through regular ¼ inch unbalanced connector.

Power for this circuit is delivered by a 12 V 5.0 ah SLA battery. This battery will be regulated by the power management system, this consists of a peak power tracker section, which regulates the current load from the 20-Watt solar cell and "trickle charge", it. This will assure that the battery is at full capacity after approximately 4.3 hours of sun exposure. The power control system will be implemented with an Arduino Mega board driving an LCD screen that will inform the user of the current charge present in the battery.

As far as the actual presentation of the amplifier goes it will be housed in an aluminum project box. This material gives the best strength vs. weight ratio for this kind of circuit.

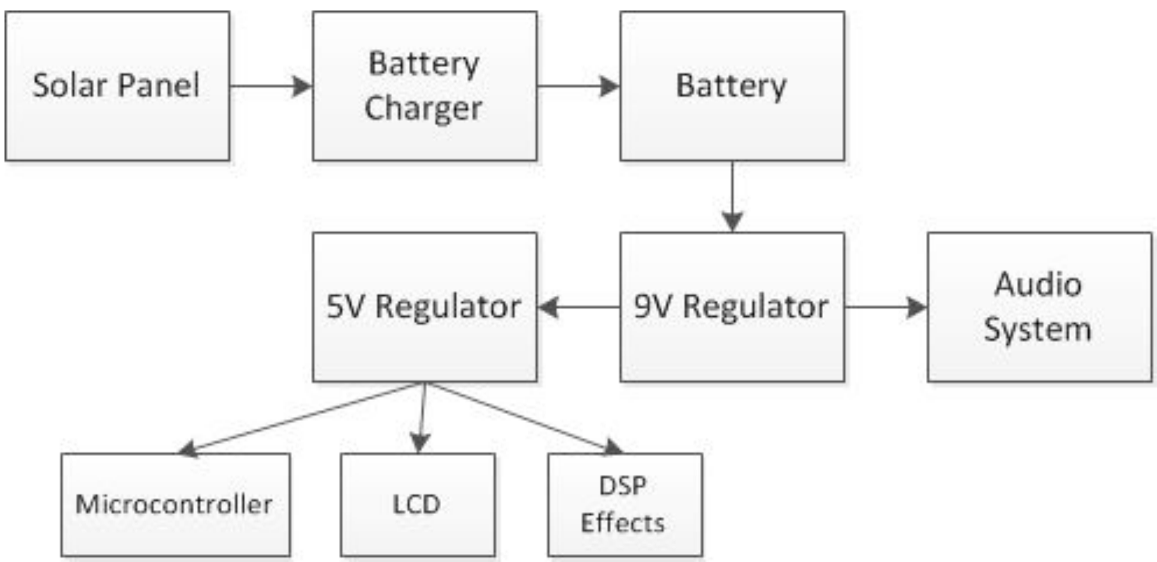


Figure 3.4.1. Overall Block Diagram

4. Prototyping

4.1 Personnel Distribution

For this project the team strived to distribute the blocks of the project as evenly as possible. Since there are three members in the group, the project was divided

into three different blocks. The first block is the power management system Gretchen Rivera is in charge of this section. This section includes many parts including a solar charger IC, solar panels, and battery. The second block is the Digital Signal Processing system. Sandra Munoz will be in charge of this system. This project block includes a DSP audio effects processor, an LCD display and a user interface to control audio effects. The last part of the project is the Audio Amplifier sub-system. Hugo Castellanos will be in charge of this section. This project block contains an audio amplifier, signal interfacing and tone control circuitry.

4.2 Audio Subsystem Prototyping

In order to test the functionality of the audio subsystem several considerations were taken. First and foremost was testing the integrity of the audio signal running through the operational amplifiers. Second, was to test that the equalizer and the audio amplifier delivered sufficient gain for it to be used outdoors. The prototype of the application circuit described in figure 3.2.5.1 and it was tested with a 4 ohm speaker load. A breadboard implementation of it and overview of the battery powered test setup can be observed in figure 4.2.1.

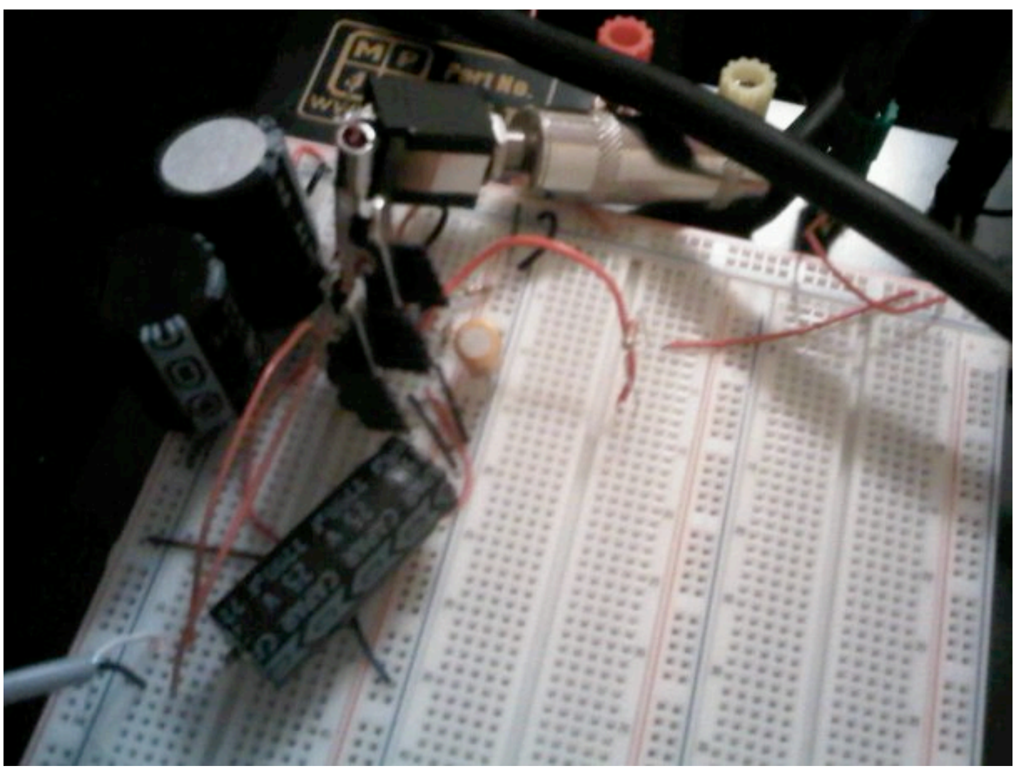


Figure 4.2.1: TDA7396 on Breadboard

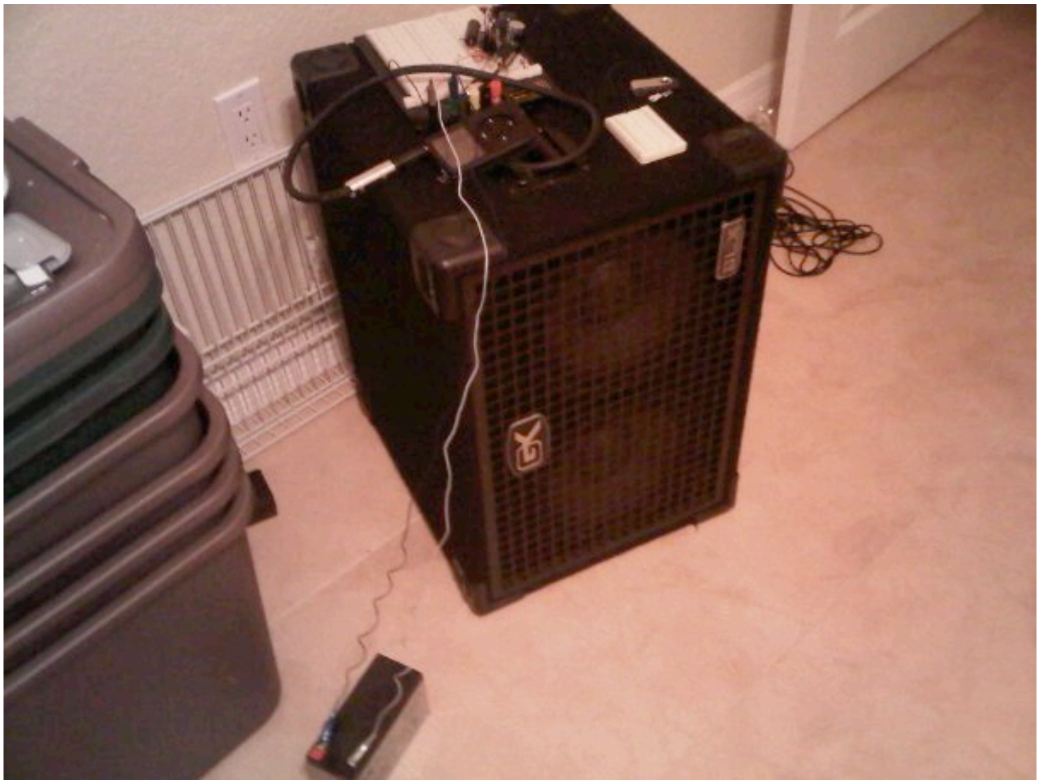


Figure 4.2.2: TDA 7396 Test Setup

Since the LME49740 IC will implement all of the essential audio functions for the Unplugged it was very important before placing it into our final PCB layout. The preamplifier was prototyped and a gain of 10 was confirmed. The unity gain mixer was tested with 3 inputs; one of 1kHz 20 mV signal generator, the same signal but running through the effects box with the "Chorus + Reverb" effect engaged and finally the iPod playing a song with a 2 Volt pk-pk swing. The output signal of the mixer was fed into the audio amplifier making sure a 10 uF capacitor was connected in series to couple both IC's together and filter the DC component of the audio signal. The test setup for said prototyping can be seen in Figure 4.2.3.

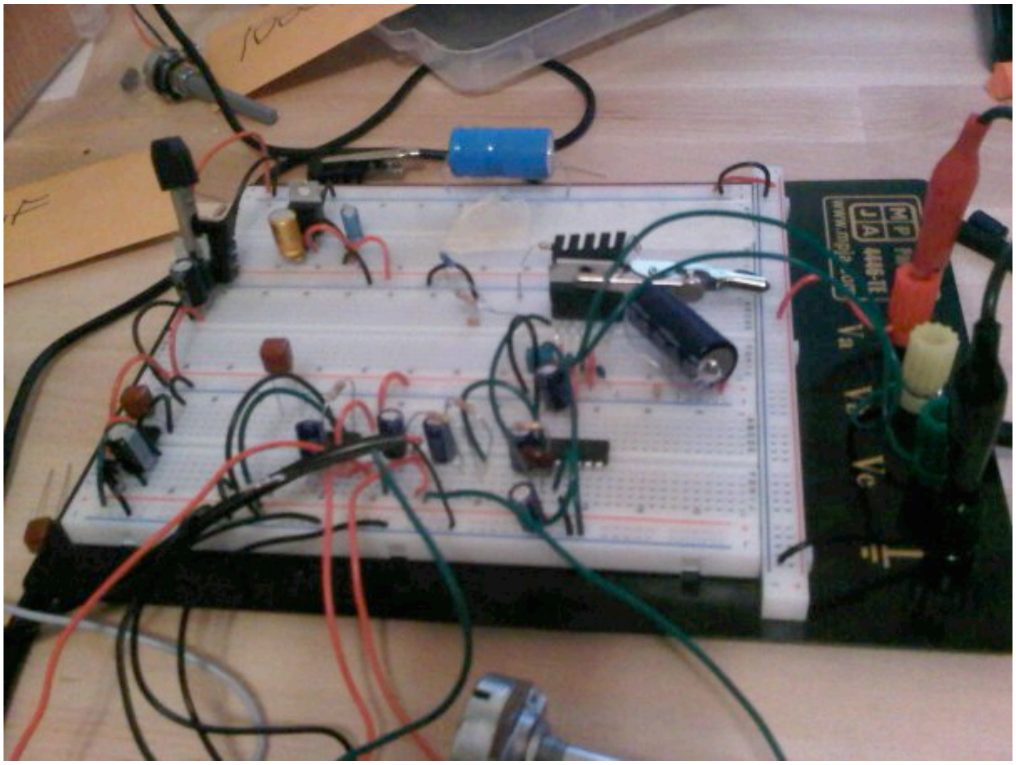


Figure 4.2.3: Mixer and Preamplifier Test Setup

Last but not least the Baxandall equalizer was prototyped and tested. A 300 Hz signal with a voltage swing of 2 V pk-pk was applied to it and the +/- 6 dB boost and cut were confirmed with the aid of an oscilloscope by varying the full range of the potentiometer from 0 kΩ all the way to 100kΩ. the same procedure was repeated for the high end of the spectrum by using a signal of 1.1 kHz and obtaining similar results with a gain of + 8 dB when the 500kΩ was fully clockwise and of -5dB when the 500 kΩ potentiometer was fully counterclockwise. A picture of the test setup can be appreciated in figure 4.1.4. All in all the prototyping of the audio subsystem proved successful. All the design requirements and goals were achieved and the subsystem was deemed to be ready for the PCB Layout.

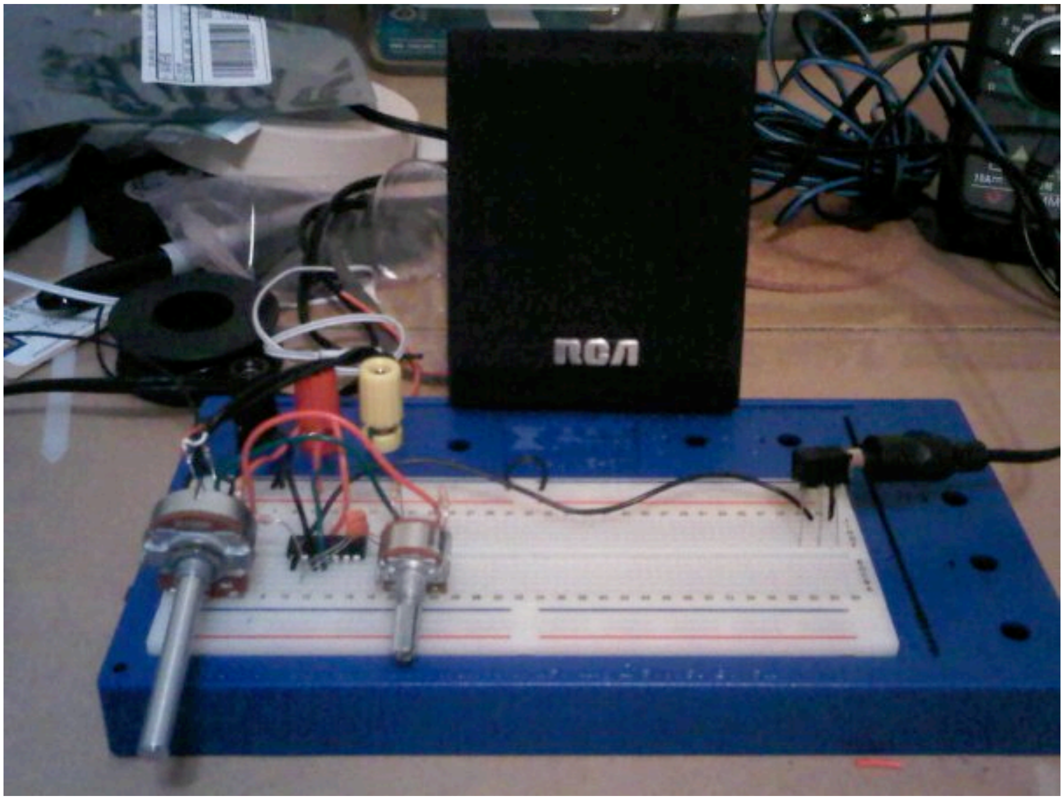


Figure 4.2.4: Baxandal Equalizer Prototype.

4.3 Power Subsystem Prototyping

To verify if the battery charger prototype was working, the solar panel was connected to the battery charger for 5 hrs charging the battery by .15V with the load disconnected. With the solar panel attached to the charger and both switches on the battery was tested with the ipod for 5hrs and it only discharged by .3V whereas it discharged by.4V without the solar panel.

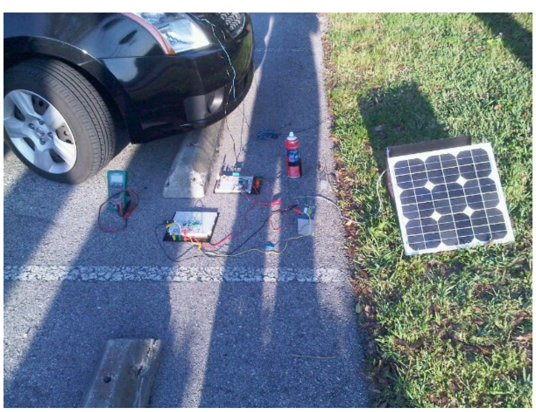


Figure 4.3.1: Power Subsystem Testing

4.4 Monitoring Subsystem Prototyping

Once the whole monitoring subsystem prototype was assembled including the solar panel, the battery and the effects unit connected to the microcontroller and the LCD, testing was run to verify that it was working correctly. To test if the microcontroller was working correctly the output voltage of the solar panel, its current and the battery voltage was measured with a multimeter. The circuit was opened in the areas where measurement was desired and the multimeter was placed to close the circuit in order to measure the current. Finally, the obtained values were compared with those displayed on the LCD display and same values were achieved. Below, the monitoring subsystem prototype can be observed.

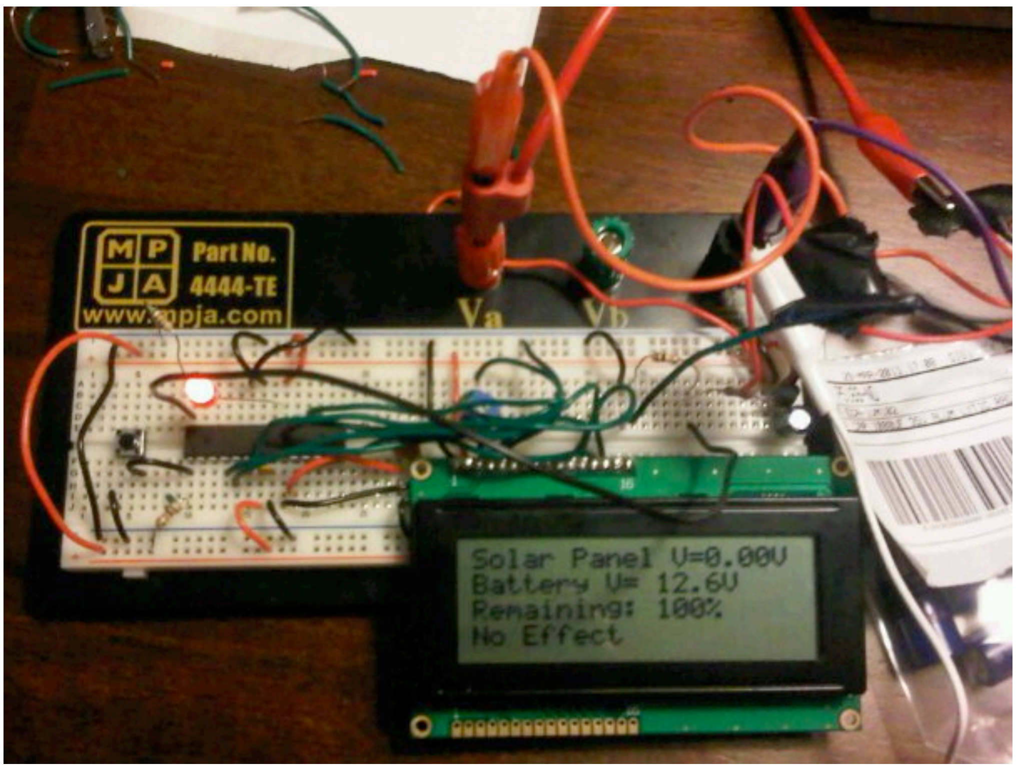


Figure 4.3.1: Monitoring Subsystem Prototype.

4.5 PCB

For the design of the PCB Diptrace software was used. Eagle was used for two weeks but the freeware version didn't have enough pins for the design of the whole system to fit in. Diptrace was used instead of Eagle because besides having enough pins for the schematic its component and pattern editor were easier to work with and provided a design library which was easy to modify. With those two programs multiple components such as the power inductor, input

diode, thermistor, and solar panel input of the battery charger as well as the LCD, DSP effects, regulators, switches and battery were designed. First the schematic of the components were sketched using the component editor with their respective sizes and then the pattern editor was used to give the components their footprints. Figure 4.5.1 shows how the schematic of the battery charger design on Diptrace looks like.

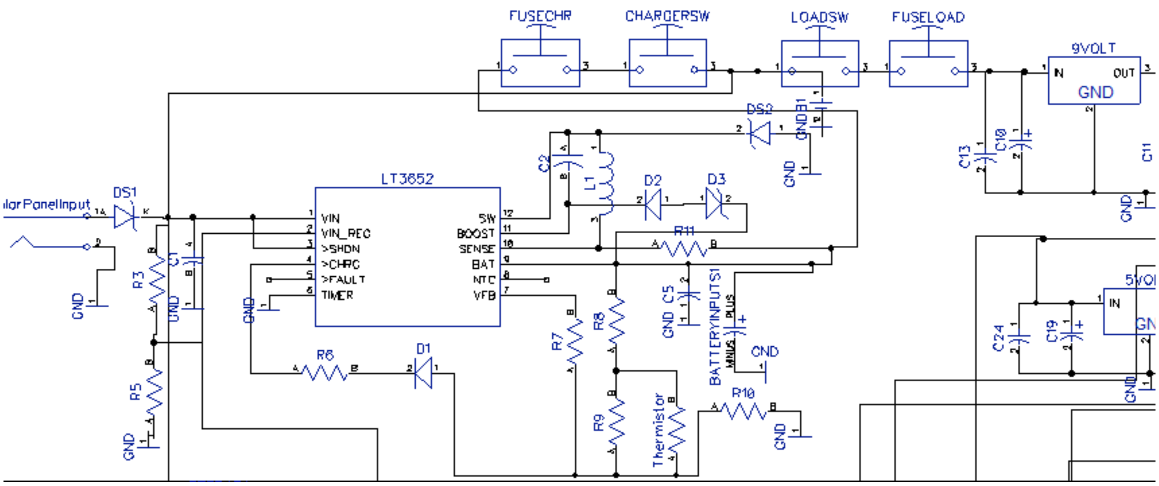


Figure 4.5.1. Battery charger and Power Distribution schematic on Diptrace.

On the other side the PCB software it comes with gives the user the ability to update the PCB with a schematic selected so the design is not lost when a schematic is changed. The PCB of the Battery charger routed is shown in figure 2 for illustration purposes.

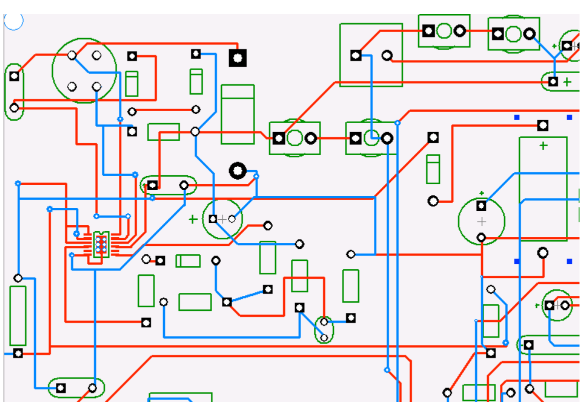


Figure 4.5.2. PCB design of battery charger and power distribution.

This PCB software gives the user the opportunity to verify the connections with the actual schematic by using the verification function named compare to

schematic. This function was very useful in the design because it highlighted if a connection was not made and showed if a bad connection was attached in the wrong place. Also, the user is able to do the routing of the PCB manually or automatically. The properties of the autorouter can be set to specific values the user desires. Another nice feature was the measure function which was used to design and measure the footprints of all components and make sure that the measurements corresponded to their respective components. All this functions can be seen on figure 4.5.3.

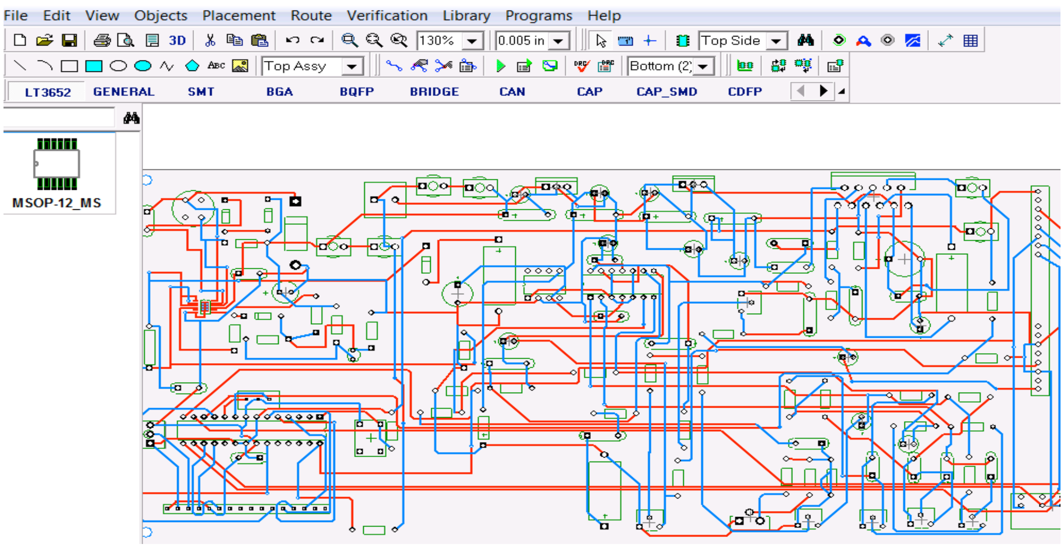


Figure 4.5.3.This figure shows the PCB design environment of Diptrace as well as all the main parts of the design together.

On the selection bar on the top, the route, placement and verification functions used can be seen. The measure function is located below the help button. After all the PCB design was done and verified with the compare with schematic function the copper pour was placed by selecting object, place copper pour, and making a border on the PCB layout. Once the border was done the user has to press enter and the place copper selection shown on figure 4.5.4 will appear. Finally, the “Connect to Net” property was set to the ground pins and the thermals were selected as four spoke to protect the PCB from overheating the components.

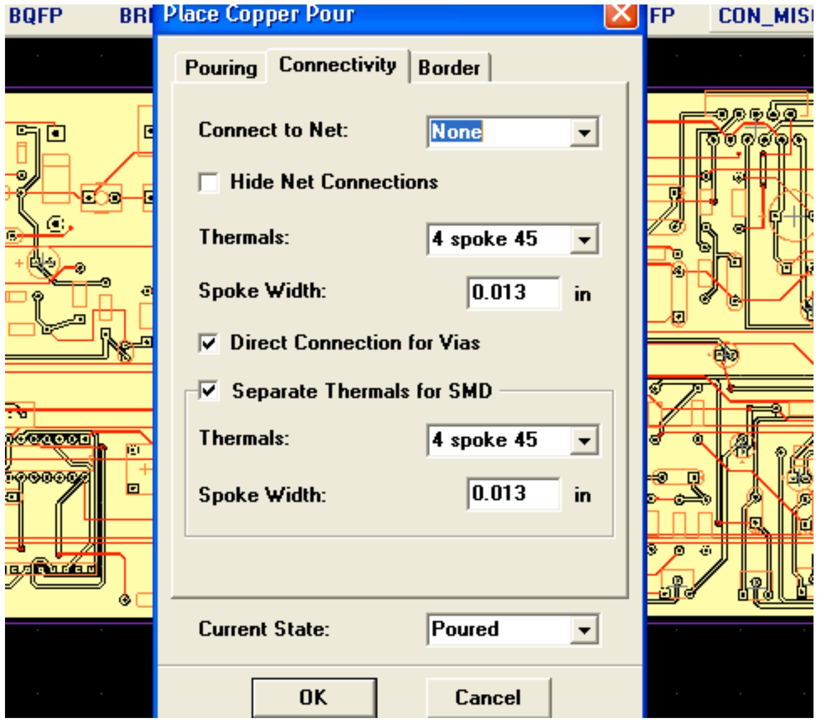


Figure 4.5.4. Place copper pour selection and its setup.

4.6 Final Testing

Once the PCB arrived and it was populated, final tests were run to make sure that every subsystem was working as expected. Below are the test results:

•A 1kHz 2mV Pk-Pk sinusoidal wave was applied to the pre-amp and a voltage gain of 10 was confirmed

•A 1kHz 1 V Pk-Pk sinusoid was applied to the audio amplifier input and a gain of 105 dB SPL was confirmed (A jackhammer 1m away is 100 dB SPL)

•The solar panel was connected to the battery charger for 5 hrs charging the battery by .15V with the load disconnected

•Battery discharged by .3V with load connected and solar panel attached for 5hrs.

•Battery discharged by .4V with load connected with the solar panel charger turned off.

• The LCD values for battery voltage and solar panel voltage were compared to multimeter measurements and they were accurate to the nearest tenth of a volt.

•The grey code encoder was cycled through every possible position and the logic values were compared with a multimeter to make sure that the LCD was displaying the correct effect.

All of the tests run produced satisfactory results and the team was happy with the final product.

4.7 Parts Suppliers

There are multiple parts that needed to be acquired for this project. The biggest concerns when selecting part suppliers were reliability, price and shipping time. Most of the parts were obtained online since it was the easiest way to get the type of parts needed for the project. Table 4.7.1 in this section shows all the parts needed fro this project and also shows the suppliers chosen to obtain the parts.

Component	Cost	Manufacturer/Distributor
20 Watt Solar Panel	$110	BP/Digi-Key
LME49740	$7.01 (3)	International Rectifier/Digi-Key
Arduino Mega Microcontroller	$59.60	Arduino/Spark-Fun Electronics
SLA 12 V 5.0 aH Battery	$21.60	PowerSonic/ Amazon.com
Belton 16-FX	$ 15	Belton Engineering LTD.
TLE2426 Rail Splitter IC	$3.40	Texas Instruments/Digi-Key
TDA7369 45 W Audio Amplifier	$14.80	ST Microelectronics
LT3562	$6.50	Linear Technology
SPST Chassis Mount Switches	$3.00 (4)	Skycraft Part Surplus
¼ inch chassis mount connector	$ 1.50 (2)	Clark Wire & Cable
PCB	$100.00	PCB Express
HD44780 LCD	$ 7.00	Ebay
4 ohm 200 W RMS Speaker	FREE	Leonardo Linares, BSM
Potentiometers	$29.53	Radioshack
Heatsinks	$9	Skycraft
Capacitors	$12.38	Newark
Breadboards	$35.80	MPJA Online
Perforated Board	Free	Hugo Castellanos
Amplifier Enclosure	$40	Ebay

Table 4.7.1: Summary of Components and Costs.

5. User Manual

5.1 Solar Charger

How to charge the battery

To charge the battery a solar panel needs to be attached to the input of the battery charger. If the solar panel is adequately attached the LCD will display the voltage of the solar panel. If the value doesn't show unplug it and connect the panel again until a value is shown. The battery charger is programmed to charge 12V lead acid batteries safely by providing the battery with a 1A constant current. To charge the battery the charger switch must be on, otherwise it won't charge. Once the current drops to .1A the charger will change to the float charge mode which will prolong battery life and avoid overcharge. If the voltage drops below 10V it will trickle charge the battery at .15A. For 12V sealed lead acid batteries do not let the battery drop to less than 11V, if this happens the LCD will notify that the battery remaining is 0% and it could permanently be damaged with acid and will no longer work. There is no way to fix this type of battery after this happens, for this reason this battery needs to be constantly charged to expand its life time.

5.2 Sound System

The *Unplugged* sound system is capable of amplyfing audio coming from musical instruments and/or mp3 players. In order to connect an instrument use a TS connector like the one shown in figure 5.2.1 and connect it to the input labeled "Instrument/Mic".

PHASE	Termination Point
POSITIVE (+)	TIP
NEGATIVE (-)	SLEEVE
SIGNAL GROUND	SLEEVE

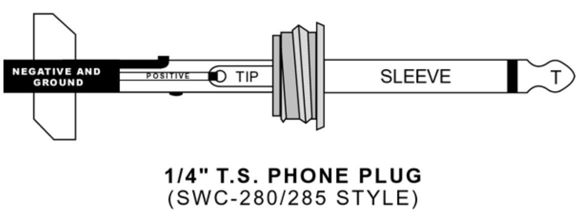


Figure 5.2.1 ¼ TS connector pin-out. 1:1 scale. Reproduced with permission from Clark Wire & Cable

To connect an MP3 player use a cable with a 1/8 in connector like the one shown in figure 6.2.2 and plug it into the port labeled "MP3".

5.2.2 1/8 TRS connector pin-out. 1:1 scale. Reproduced with permission from Clark Wire & Cable

5.3 Device Operation

The unplugged's faceplate has the following layout shown in figure 5.3

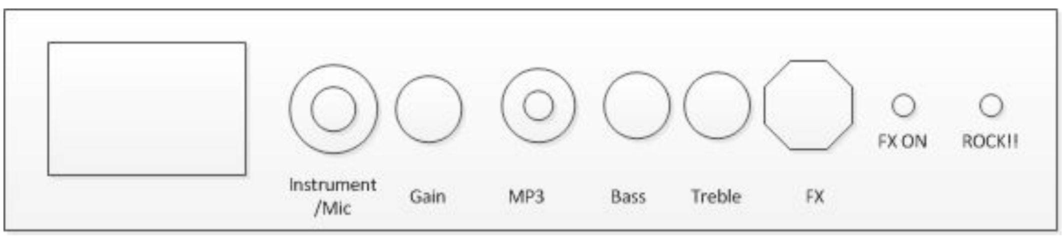


Figure 5.3.1: Front view of device

The different buttons perform the following functions:

Gain: Affects only the instrument/Mic input. Turn it clockwise to increase the volume of the microphone or guitar plugged into the device.

Bass: Affects only the device plugged into the MP3 input. When turned counterclockwise the knob will reduce the bass response of all audio at 320 Hz up to 6 dB. When turned fully clockwise it will boost the bass response of all audio up to 6 dB.

Treble: Affects only the device plugged into the MP3 input. When turned counterclockwise the knob will reduce treble response of all audio at 1.1 kHz up to 6 dB. When turned fully clockwise it will boost the treble response of all audio up to 6 dB.

FX ON: This knob turns on the built-in 16 effects DSP processor inside the *Unplugged.* The audio effect only affects the audio device plugged into the Instrument/Mic input of the device. See FX button.

FX: This knob possesses 16 possible positions. Whenever the FX ON button is on the on position the audio coming in through the Instrument/Mic input of the device will be affected with any of the following effects:

01 Hall 2.0 Seconds	05 Room 1.0 Seconds	09 Delay 125 ms	13 Chorus + Room
02 Hall 3.0 Seconds	06 Plate 3.0 Seconds	10 Delay 200 ms	14 Chorus + Echo
03 Room 1.6 Seconds	07 Plate 4.0 Seconds	11 Chorus Slow	15 Flanger Slow
04 Room 2.0 Seconds	08 Plate 1.0 Seconds	12 Chorus + Hall	16 Flanger Fast

Table 5.3.1: Effects implemented by the *Unplugged*

These effects are designed to make sure that a microphone used in a public address situation truly stands out. Also they sound really good when used with a Guitar, Bass or any other Instrument.

ROCK!!: "When feeling an urge to melt faces of people around you plug in your weapon of choice (Bass, Guitar or Microphone) into the Instrument/Mic input and turn the "Gain" knob up to 11. Proceed to turn the "ROCK!!" button into the "On" position and engage in face melting activities

5.4 Monitoring Subsystem

The *Unplugged* is equipped with an LCD display that provides the user with the following information:

Solar Voltage: States how much voltage is the device getting from the sun. anything above 17.2 V will start to charge the battery.

Battery Voltage: states how much voltage is the device getting from the battery. Max is 14.4. Min is 11.0 Volts

Remaining: Displayed as a percentage. It lets the user know what is the percentage of the battery life left.

Effect: if the “FX ON” button is in the off position it will display a “No Effect” message. Otherwise it will display to the user wich audio effect is engaged in the instrument/mic input of the device.

5.5 Troubleshooting

No Audio:

1) Check battery connections. If connected move to 2)
2) Check audio connections. If connected move to 3)
3) Check “ON” button if engaged move to 4)
4) Check gain level. If above 0 move to 5)
5) Check connection from speaker output to actual speaker. If connected move to 6)
6) Contact customer service at (407) 277 9064

LCD display not on:

1) Check battery connections. If connected move to 2)
2) Check “On” button. If engaged move to 3)
3) Push black “reset” button to the right of Atmel 328 microcontroller.
4) Adjust blue contrast knob located to the left of Atmel 328 microcontroller until displays behaves properly.

6 Milestones and Financing

6.1 Budget and Financing

The financing for this project was fully provided by Hugo Castellanos. The only condition for this is that he will keep the project once it was fully built. The maximum amount that Hugo was willing to provide for this project was $1,000.00. On table 6.1.1 specifies the final budget for this project specifying the cost of every part that was included. It can be observed from this table that eve though the original budget was $1000, the final budget for all the parts was $573.54 which was very under budget.

Item	Price
Solar Panel	$ 108.95
Arduino Mega	$ 59.65
Battery	$ 21.60
IC's	$ 44.63
DSP FX	$ 15.00
LCD	$ 7.00
Heatsinks	$ 9.00
Capacitors	$ 12.38
Breadboards	$ 35.80
Potentiometers	$ 29.53
Perforated board	Free
Mis. Audio Connectors	$ 40
Amplifier enclosure	$40
4ohm 15 inch speaker	Free
PCB	$ 150
Total Spent	**$ 573.54**
Original Budget	**$ 1000**

Table 6.1.1: Total Spent

6.2 Milestone Chart

Milestones were very important in order for the team to have a game plan and stay in schedule . For this project, milestones were set up for specific times and specific group members. The milestone chart on table 1 was a very important tool as the group members used it to track progress and to make sure that the

respective responsibilities were met. The milestones described in this table were for two semesters, senior design 1, and 2.

Date	Person in Charge	Milestone
Fall/10		
9/24/10 – 9/30/10	Hugo	Research in sound engineering
	Gretchen	Research in Solar Panel and battery implementation
10/11/10	Sandra	Research DSP Implementation research
	Gretchen	Research Solar Panel and Battery implementation
	Hugo	Research Audio Amplifier implementation
10/18/10 - 10/31/10	All Members	Preliminary Research on possible ways of implementing all 3 subsystems.
November – (4 weeks)	All Members	Paper Writing
		Paper Revising
		Device Search
		Vendor Search
December – (2 weeks)	All Members	Prototype Preparation
		Start ordering components (specialized op-amps, capacitors, PCB board, etc)
Spring/11		
January – (4 weeks)	All Members	Audio Sub-system prototyping/laboratory Testing
		DSP Sub-system prototyping/laboratory testing
		Power sub-system prototyping/laboratory testing
		“Laboratory testing” will consist in: - Testing circuitry for continuity, energy performance and power consumption. - Testing logic and microcontrollers for functionality.

		- Testing audio amplifier, parametric equalizers for frequency response
Date	**Person in Charge**	**Milestone**
February- March (8 weeks)	All Members	Audio subsystem Field Testing
		Power Subsystem Field Testing
		DSP System Field Testing
April – (4 weeks)	All Members	Start looking for EECS faculty to be part of our final presentation panel.
		Revise all documentation saved during Senior Design 1 and 2 (videos, photos, schematics, field test results) and put it together for the final presentation
		Rehearse and Time our final presentation.
		Final Testing of the complete System.

Table 6.2.1: Milestone Chart

7. Summary and Conclusion

This project was inspired by love of music and solar power. The concept came together as the group discovered that there are not any products out in the market that allow an amplifier to run on battery power but also are able to obtain all the energy that they need from the sun. What sets the *Unplugged* apart from other devices is the fact that it does not depend from mains power at all. The user is free to take the whole system anywhere and plug in a portable audio player, a laptop, a guitar or a microphone into it. The fact that the device has a built-in DSP effects processor enhances its functionality as a public address system as well as a portable amplifier for guitars. Said effect can be easily observed in the incorporated LCD display. Although all of the aforementioned audio sources are supported the *Unplugged* really shines whenever it is used with an MP3 player. Testing showed that audio can be reproduced up to 105 dB SPL. For reference, a jackhammer 1 meter away from you delivers 100 dB SPL. This assures that the *Unplugged* sound system is powerful enough to be used in a variety of outdoor events without the need of running cumbersome extension cords. As far as power consumption is concerned the final product is ready to provide at least 3 hours of distortion-free music at the levels mentioned above. Any concern as far as battery levels and can be easily observed by the user in the incorporated LCD display. All in all the *Unplugged* is ready to provide hours of entertainment for any outdoor event.

8. Appendices

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1Wayne Storr from electronics-tutorials.ws

Hello Hugo,

Firstly, thank you for you email and for asking in advance to use some of my schematics for your Engineering project. Most people would have just copied them regardless.

As you have kindly asked and your class is Electronics related, I would have no objection to you using the content or graphics from some of my tutorials. However, I must ask that you reference my work and site www.electronics-tutorials.ws accordingly.

Good luck with your Electronics project.

Kind Regards.

Wayne Storr
webmaster@electronics-tutorials.ws

----- Original Message ----- From: "Hugo Castellanos" <hugocastellanos1@mac.com>
To: <webmaster@electronics-tutorials.ws>
Sent: Saturday, December 04, 2010 10:41 AM
Subject: User Question

Fellow Electronics Enthusiast,

I am an electrical engineering student and I am building a solar powered audio amplifier for my senior design project. I would like to ask you permission to include the schematics for class B and Class A electronic amplifiers in my final paper. Thank you beforehand.

Respectfully,

Hugo Castellanos
hugocastellanos1@mac.com
(407) 435 4234

2) Clark Wire & Cable

Hugo,

Thank you for checking. Yes, you have our permission.

Good luck with your project!

Scott

On 11/29/10 2:27 PM, "Hugo Castellanos Bermudez" <hugocastellanos1@mac.com> wrote:

Dear Audio Professional,

I am an electrical engineering student and I am designing a portable audio amplifier for my senior design project. I would like to request your permission to reproduce the connector pin-out images for the TRS and XLR balanced connectors in my final report. I would of course give credit to the company for this. Thank you in advance.

Hugo Castellanos
(407) 435 4234
hugocastellanos1@mac.com

Scott Fehl
Product Marketing Manager
Clark Wire & Cable

408 Washington Blvd.
Mundelein, IL 60060
Toll Free: 800-CABLE-IT
Local: 847-949-9944
Fax: 847-949-9595
Mobile: 630-989-7227
scott.fehl@clarkwire.com
www.clarkwire.com

3) NXT Permission letter

From: tac@nxt.com
Subject: Image Usage Request [Incident: 101204-000017]
Date: December 5, 2010 8:22:42 PM EST
To: hugocastellanos1@mac.com
Reply-To: tac@nxt.com

Recently you requested personal assistance from our on-line support center. Below is a summary of your request and our response.

If this issue is not resolved to your satisfaction, you may reopen it within the next 7 days.

Thank you for allowing us to be of service to you.

To update this question by email, please reply to this message. Because your reply will be automatically processed, you MUST enter your reply in the space below. Text entered into any other part of this message will be discarded.

[===> Please enter your reply below this line <===]

[===> Please enter your reply above this line <===]

4) Malcolm J. Jones from PowerSonic

Dear Gretchen,

Thank you for your e-mail. By all means, please use whatever materials you find useful.

I am attaching an image of the battery concerned and two Power Points –just in case you might find them handy.

If you need any further information you know where to find me!

Best of luck with the paper.

With kindest regards,

Malcolm J Jones
Marketing Director
Power-Sonic Corporation
7550 Panasonic Way
San Diego, CA92154
Tel: 619-661-2020
Fax: 619-661-3650
Website: www.power-sonic.com

We've Got The Power!

From: Javad Alabadi [mailto:javad@power-sonic.com]
Sent: Wednesday, April 13, 2011 9:13 AM
To: mjones@power-sonic.com
Subject: FW: SLA PS 1250 Battery Datasheet

From: gretchen rivera [mailto:gretchen.rivera@knights.ucf.edu]
Sent: Tuesday, April 12, 2011 5:59 PM
To: technical-support@power-sonic.com
Subject: SLA PS 1250 Battery Datasheet

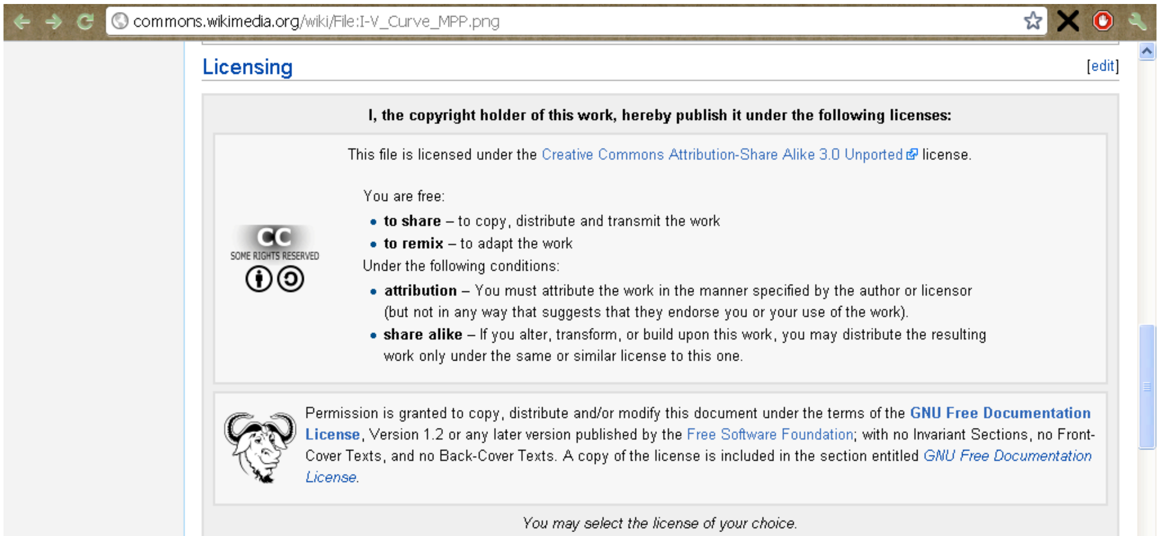
Hi,

I am working on my senior design project for the University of Central Florida and I am using the PS 1250 battery for a part of my design. I would like to use part of the datasheet to show the capabilities of the battery on my final paper. I will add the information of the datasheet on my reference part. Would the company give me permission to do that?

Thanks,

Gretchen M. Rivera
Electrical Engineering
University of Central Florida

5) License used for Figure 3.2.1.3: Example of a Solar Panel IV Curve with MPP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Regards,

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office: 480-792-4391
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From: Sandra Munoz <samu224@gmail.com>@MICROCHIP **Sent:** Friday, December 03, 2010 2:17 PM **To:** University **Subject:** Permission to use image

Hi, My name is Sandra Munoz. I currently attend the University of Central Florida and I am working on a Senior Design Project. I would like to include one of the pictures of the pic18FJ90 block diagram on my senior design documentation. I want to ask you permission to do so. The section that It will be included on is on research of components to include on the project. Thank you, Sandra